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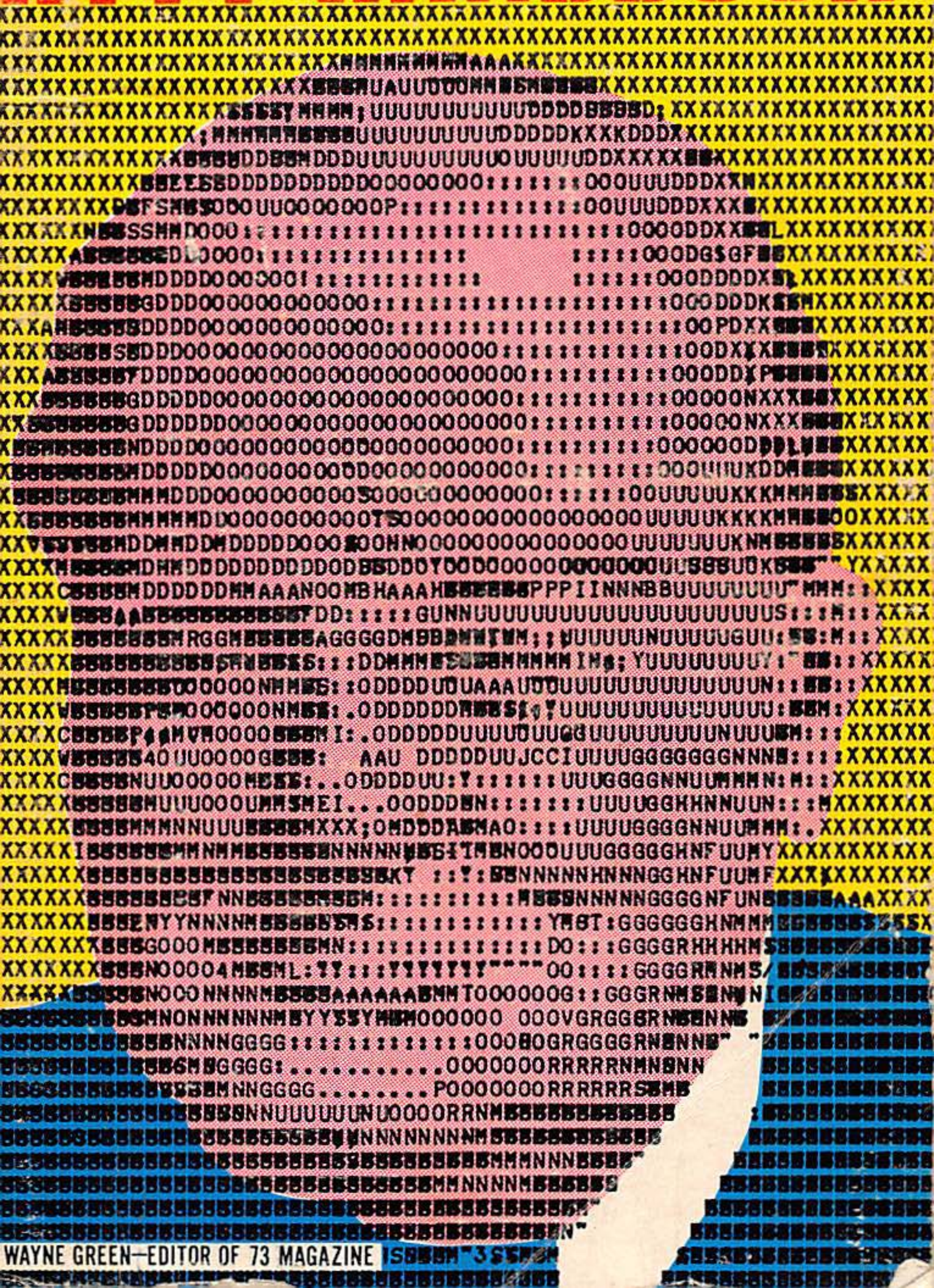
RTTY HANDBOOK



WAYNE GREEN—EDITOR OF 73 MAGAZINE

RTTY HANDBOOK

RTTY HANDBOOK

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TAB BOOKS
Blue Ridge Summit, Pa. 17214

FIRST EDITION

FIRST PRINTING—MAY 1972
SECOND PRINTING—JANUARY 1974

Printed in the United States
of America

International Standard Book No. 0-8306-2597-6

International Standard Book No. 0-8306-1597-X

Library of Congress Card Number: 71-189957

RTTY HANDBOOK

PREFACE

The symbol F-1 on authorized emission charts in FCC regulations may not mean much to some amateurs. To others it is an open door to an entirely different concept of communications. It signifies a technology which is challenging enough for the most learned engineer, yet basically simple enough for the greenest neophyte. F-1, along with the symbols A-1 and A-2 authorizes amateur radio-teletype (commonly called RTTY) on all bands except 160 meters.

Teletype is actually the registered trademark of the Teletype Corp., the leading manufacturer of teleprinter machines. Over the years teletype has come to be a term used synonymously with teleprinter, which is the actual descriptive name of the machine.

"Why amateur RTTY," when we have CW, AM and SSB available to communicate with other amateurs in most countries throughout the world. The answer is simple. Amateur radio, by international agreement, is designed as a service dedicated to furthering communications technology all over the world. It is a means of bringing together people from all walks of life, all races, and religions who share a common interest, i.e., to communicate ideas and thoughts to each other. Amateur radio is also dedicated to furthering technical knowledge of many different means of communication by encouraging experimenting in these areas. This is commonly called advancing the state of the art.

Amateur radio is made up of thousands of hardy individuals who are looking for a challenge or project just a little bit more difficult than the one they have just been involved in. These people, although not forsaking the old tried and proven ideas and modes of communication, are interested in new and different means of communicating. It is these people who look to the different areas of amateur activity such as amateur television, facsimile, slow scan TV, FM, narrow band FM, and amateur RTTY for their enjoyment.

Wayne Green

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CHAPTER 1

Getting Started in RTTY

For many years, commercial telephone and telegraph companies have used teletype to transmit and receive printed messages from point to point in the U.S. and other countries. Teletype is a means of transmitting signals through wires from one teletype machine to another. A DC voltage is switched on and off in a coded pattern which the receiving machine understands and converts to mechanical motion. This causes type characters to strike a sheet of paper, thereby printing the message in directly readable form (Fig. 1-1).

CQ, CQ, CQ

Amateur radio operators saw the opportunity to utilize another communication media when these machines came

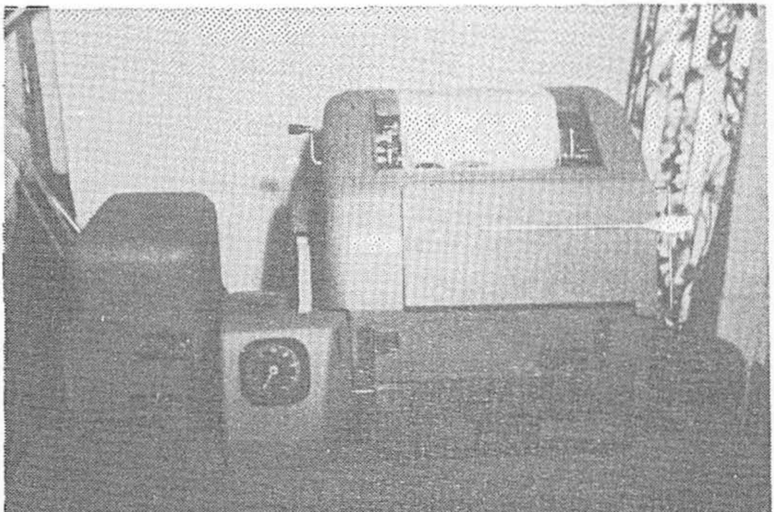


Fig. 1-1. Model 19 Teletype machine. Tape transmitter is on left, end of line indicate dial in center, perforator behind dial.

into use. The first problem they had to overcome was how to get the pulsing code signals from one machine to another. After trying various methods which worked crudely and only at times, these pioneers developed the process now called frequency shift keying (FSK), or F-1 as termed by the FCC.

Mark and Space

The FSK transmitting process is basically this: The transmitter is on the air and transmitting steadily at a given frequency called the "mark" frequency. When the desired character key on the teletype keyboard is depressed, a motor driven mechanism produces a switching off and on of a DC voltage in a code group assigned to the desired character. This voltage then activates an electronic unit which causes the transmitter's frequency generating section to shift the transmitted frequency very slightly for a period of about 22 milliseconds. It is this short period of frequency shift, transmitted in code groups of the proper pattern, that carries the teletype signal over the air. The frequency shift may be any value up to 900 Hz as allowed by FCC regulations. The most commonly used shift is 850 Hz, although shifts as low as 170 Hz and 160 Hz are being used occasionally. This shifted frequency (generating the mark frequency minus 850 Hz) is called the "space" frequency.

A variation of the FSK which is authorized on high bands only (6 meters and up) is AFSK which means audio frequency shift keying. This method utilizes the constant carrier on action of either AM or FM; but, instead of a voice signal being transmitted, an audio tone of 2125 Hz is generated in an audio signal generator and shifted by the teletype machine 850 Hz to a tone of 2975 Hz. This audio signal is impressed upon the carrier frequency in the same manner that a normal voice signal would be transmitted. This system does not involve changing of the basic transmitting frequency as does the FSK system.

Incoming Signals

The receiving of RTTY signals requires receiving equipment of reasonable stability and selectivity. The incoming FSK signal is treated the same as a CW signal in that it is detected and changed to an audio signal by the bfo control on the receiver. The audio output to the speaker is tapped and some of this signal is fed to the RTTY converter. The converter amplifies these signals slightly and feeds the complete

audio signal through two filters which allow only audio tones of a certain frequency to pass through them. Usually these two audio frequencies are 2125 Hz and 2975 Hz. The bfo on the receiver is tuned until the steady audio tone coming from the receiver is 2975 Hz (mark frequency). This audio tone will pass through the mark filter. The frequency shift of 850 Hz will change the audio tone to 2125 Hz (the "space" frequency). This tone will pass through the "space" filter on the converter. These two audio tones are then amplified further and one or the other becomes the triggering pulse for an electronic switch which switches the DC voltage off and on in the proper sequence to operate the teletype printing mechanism. The combination of off-on pulses fed to the printing unit sets up the mechanical reaction necessary to print the proper character on the paper.

Equipment

The equipment involved need not be expensive. In fact, by using an existing AM-CW transmitter (either crystal or vfo controlled) and the station receiver you already have, you can be on the air on RTTY for less than the cost of most SSB transmitters on the market today. The teletype machine itself will probably be the most expensive investment, costing anywhere from $20.00 and up. The RTTY converter can be either purchased commercially or home built. The cost of this unit could run anywhere from $15.00 and up for a home built unit, to $200.00 for a commercially built version. The FSK unit generally must be designed and built to match whatever transmitter you are planning on using.

Amateur RTTY is a tremendously fascinating field. It has its problems and frustrations just like the other modes; but that is the real test of an amateur's dedication to the goal he pursues.

IS RTTY FOR ME?

There are several questions that are frequently asked by the amateur who would like to give RTTY a try: "Is it very expensive?" "Is it very technical?" and "How can I get the equipment?"

There has been a great deal written about all three of these questions and the answers that stop most amateurs are those that begin with "RTTY is expensive but..."' or "a degree of technical skill is required..." Many articles that use these lines for leads, do more to turn amateurs away from RTTY, than any other thing.

The fact is that "going RTTY" is one of the lowest cost layouts that an amateur can own. The average amateur has a considerable sum invested in a good commercial transmitter and receiver, plus the auxiliary equipment he has acquired. These items run as a rule to almost $1,500 per amateur station; so, if we use this as a yardstick, we find that adding RTTY is a rather minor additional expense compared to what he has already invested.

Let us take the question "Is it expensive?" and see just how much it does cost to get on RTTY.

Is It Expensive?

One amateur's first installation consisted of a model 26 machine and table which cost $50, a home built terminal unit (parts taken from his junk box—none too close to schematic specifications) which cost, as a result, about $5. If you had to purchase most of the parts, it would not be over $30, and the FSK circuit made from old spare parts cost nothing. If the parts were purchased today, they would cost a couple of dollars.

Quick addition will show then that this amateur went RTTY for about $55 plus a little construction time. Of course, it is assumed that the transmitter and receiver are already part of the station, so they cannot be counted into the total cost of getting on RTTY.

Model 26 machines are still available from time to time for under fifty dollars, and terminal units can also be purchased for a very low figure or built from spare parts. The expenses involved will depend upon the interest of the individual who demands more equipment, as his interest grows. This is true of any hobby and should not be used as the basis for the concept that RTTY is "too expensive."

Is It Very Technical?

You have a receiver and you have a transmitter, and you know how to use both, and you know that this does not require a great deal of technical skill. Now let us apply it to receiving and sending RTTY signals.

First let us look at the receiving side. Besides the receiver, you need a terminal unit (converter) and a teletypewriter. Since you are familiar with tuning a CW or SSB signal which requires your being familiar with bfo adjustment, you should have no trouble tuning a RTTY signal which is nothing more than a steady carrier, being shifted in frequency 850 Hz, by the

transmitting station. This produces a two tone audio signal from the receiver output and feeds this to the terminal unit, which converts these audio signals into DC pulses which then are fed to the machine, operating the selector magnets on the machine, producing the printed characters being transmitted.

If you can connect equipment together in proper sequence, you will find nothing complicated about it. And, if you can build simple circuits, it will be a cinch.

Now let us take the transmitting end of the story. A lot will depend on the type of transmitter you have, but nearly all can be made to operate RTTY with FSK or AFSK. Take, for example, an old Navy TCS transmitter. The only thing needed was to run three wires from the base of the oscillator tube. One went to the cathode and the others were to pick up filament voltage for the 12H6 diode used. That is all there was to it.

The FSK circuit was simple, consisting of a 50,000 ohm pot, a 2.5 mh RF choke, two .005 condensers, an octal socket, the 12H6 tube, and a small slugged tuned coil form ½ inch in diameter, wrapped with 20 turns of No. 22 wire. So you will understand that it does not take a radio engineer to do it.

Feeding this frequency shift keyer, to the cathode of the oscillator causes the frequency to be shifted, according to the amount of inductance inserted into the oscillator circuit.

The adjustment of the 50,000 ohm pot in this particular circuit governs the amount of shift and should be set up for 850 Hz.

That is all there is to it, except to add, that the answer to "How can I get the equipment?" is to contact any RTTYer near you or, better yet, join one of the RTTY societies, who will be glad to assist you in getting machines and advice.

SECURING A PRINTER

There are estimated to be about 8000 amateur radio stations equipped for radioteletype operation. This is probably a low estimate. You can tune in on these stations and hear them give their call signs on CW after each transmission if you check 3620 or 7040 kHz, or 146.70 MHz on 2 meters.

What is there that makes an amateur go to all the trouble of locating a printer, shelling out the cash, building or buying a converter, and setting up RTTY shop? Who knows? But here is how it is done.

There was a time when the greatest problem to be solved in getting on RTTY was the procurement of the machine. This sometimes used to take months. The result of this waiting was an ancient Model 12 machine. All of the machines had to come

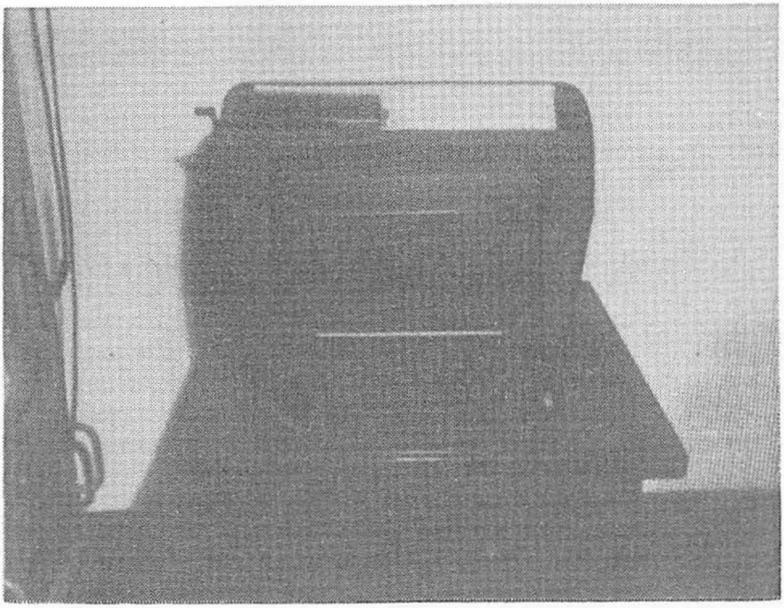


Fig. 1-2. The Model 26 printer.

through the operator of W2BFD back in those days, since he was the one who set up the only method of liberating them from the Bell Company and Western Union. Now several outlets are working and machines are no longer in scarce supply.

Model 12

There are still a number of the old Model 12 machines being used. They are almost indestructible. The main problem with them is that they generated more electrical noise than the more modern machines. They have six magnets in the typing unit and each one requires 300 ma to flip. The polar relay made quite a bit of hash interrupting all this current. Good bypassing or the substitution of tubes or transistors to operate the magnets calms things down to usefulness.

Model 26

It is much simpler with the later machines though, since they use just one magnet and that draws only 60 ma. The Model 26 (Fig. 1-2) was released when they were found to be

unsatisfactory for 24 hours-a-day news operation. They are very good for ham work, but are a bit light to leave unattended for weeks on end. This model, like the old Model 12, is a page printer where the paper moves past a fixed typing unit, like a typewriter. The Model 12 has bails that come up and hit the page, also like a typewriter. The 26 has a small type wheel that spins to the right letter and then strikes it against the paper, something like the kiddy typewriters.

Model 15

The Model 15 machine is the most often seen. It is still standard for most uses. You'll see these at newspapers, radio stations, TV stations, offices, etc. They are now being replaced with the Model 28. The paper stays still in the 15 and the type basket moves back and forth in front of it, pecking away.

Model 19

The Model 19 is a Model 15 printer with a set of tape equipment. These are nice, do not pass them up if you get a chance to get one reasonably.

Model 14

The Model 14 is a strip printer. While page printers are more in vogue among ham TT'ers, there is much to be said for the strip printer where you can peck away and not worry about when you are getting to the end of the line, unless you happen to be in contact with a fellow with a page printer, in which case you normally do have an end-of-line indicator built into your 14. Some Model 14's have a tape puncher built in. Very good. Once you get to playing with tape you will have even more fun.

Model 28

The Model 28 (Fig. 1-3) is the newest printer in general use. The price is very high on these and few have found their way into amateur hands. They will print at 60, 75, or 100 wpm. The paper stays still again and the type box glides back and forth in front of it, moving into position and then printing the selected letter.

Model 31A

The Model 31A (Fig. 1-4) is a new portable printer, using $5/32$ in. strip paper. It is only 20 pounds and very small. This

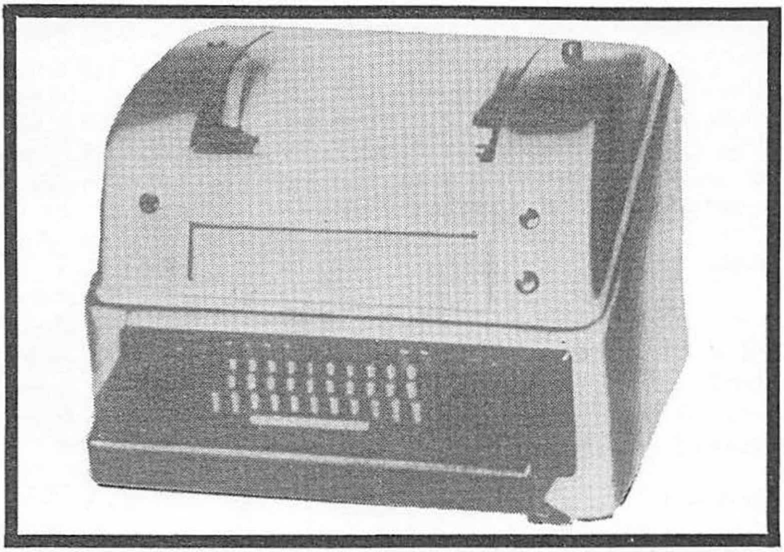


Fig. 1-3. Teletype Corporation Model 28 printer set. This is the modern replacement for the Model 15.

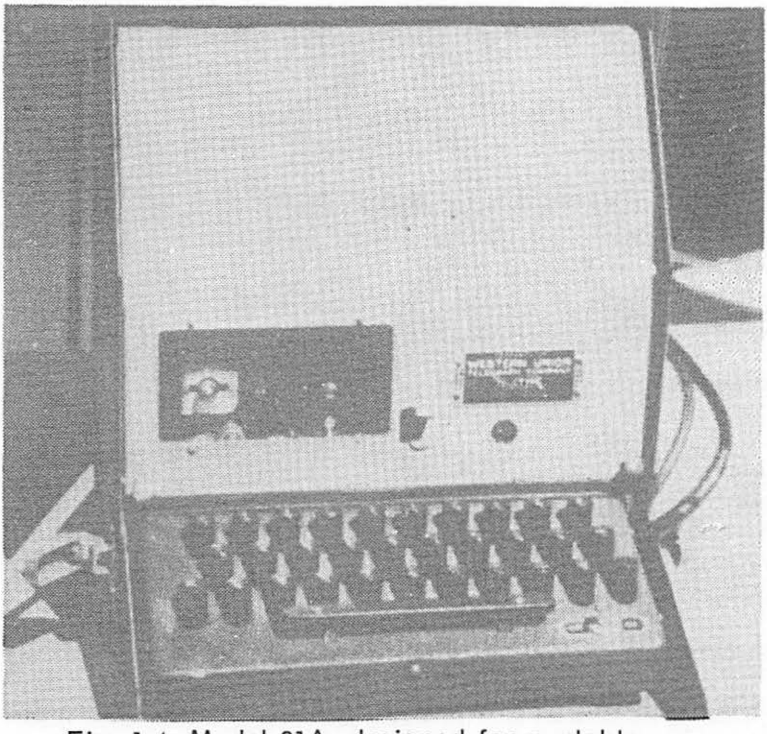


Fig. 1-4. Model 31A, designed for portable use.

one also has an end-of-line indicator for use when contacting page printers. Most printers require 100 volts DC to operate the magnets, this one needs only 26 volts DC.

You can best solve the problem of locating a machine for purchase by contacting your nearest radioteletype society and taking their suggestions. The printer will probably cost in the neighborhood of $75, though you can spend a lot more if you wish. Most operators use page printers which print on long sheets of 8½-inch wide paper. This paper comes in rolls or in cartons fanfolded and is easy and inexpensive to get either way. A few operators prefer the strip-type printers, but these are in the minority. Get a page printer for your first unit (Fig. 1-5).

Once you have the printer you will naturally want to sit right down and start typing to see how it works. You have to supply 100 volts DC to it in order to get any action. Most machines have several cables coming out with phone-jack plugs on the end. Hook the 110 volts DC between the plug tips. Plug in the AC cord, also.

Next, you will want to start trying to copy some of the RTTY signals coming in over the air. This takes a little more preparation. To accomplish this you have to turn the "beedle-beedle" signals coming out of the receiver into DC pulses. Here the converter or terminal unit becomes necessary. But all this in due course. Before we proceed, let us discuss something of the development of these devices.

THE EARLY DAYS OF RTTY

In the beginning of amateur RTTY, there was only one type of converter: this was the W2BFD unit, which consisted of two stages of tuned audio amplification, a rectifier to change this to DC, a DC amplifier, and a polarized relay. The early days of RTTY were spent up on two meters, since that was about the only band where any type of frequency shift keying was permitted. A few tried the old eleven meter band and experimented with straight frequency shift keying as well as the more popular audio frequency shift keying (AFSK). After a long battle, the FCC granted permission to use FSK on the lower ham CW bands. This was when RTTY came of age.

John Williams, W2BFD (Fig. 1-6), who was largely responsible for getting amateur RTTY started chose the commercial standards of 2125 and 2975 Hz for the two audio tones to be used. His amplifiers were tuned to these frequencies. Note that the two are 850 Hz apart. A lot of thought and design went into that original choice and it is still

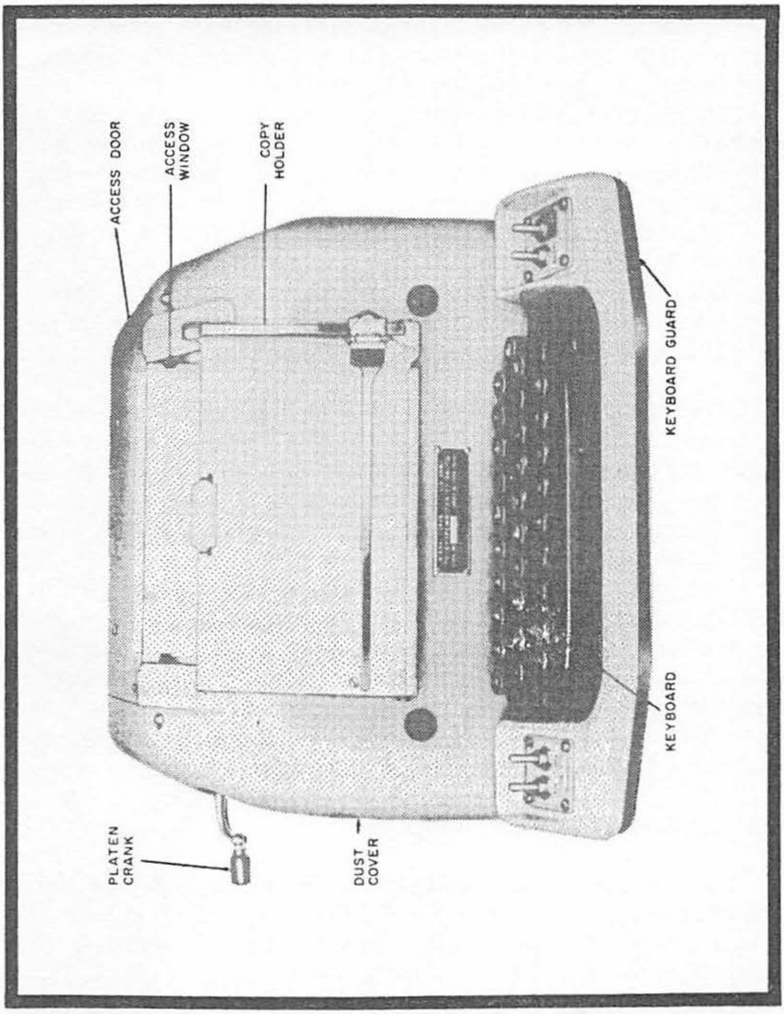


Fig. 1-5. Teletypewriter set AN-FGC-20. This Kleinschmidt printer is made for the Armed Forces.

the standard that we are using in spite of many attempts to improve on it.

FREQUENCY SHIFT KEYING

A lot of experimenters down through the years have tried many systems of sending RTTY via radio, and every time they

decide that frequency shift keying (FSK) is the best method. In the early days of RTTY, just after WWII, FSK was not permitted on the lower frequencies with the result that few TT stations operated on these bands. It was actually possible to send RTTY signals with make-break (CW) keying, but the received signal had to be strong and clear of interference to give perfect copy. With FSK it is possible to demonstrate perfect copy with a signal that is not audible to the ear.

FSK, or F-1, emission consists of sending a continuous carrier and letting the keying shift it in frequency rather than turn it on and off. The frequency shift that we have accepted as standard is 850 Hz. If you tune zero-beat with any RTTY station you will hear an 850 Hz note from the other frequency he is sending.

Transmitting

Most amateur stations remotely key a small capacity across the vfo circuit to change it by the desired 850 Hz. This is very simple to do. It is also simple to adjust the amount of

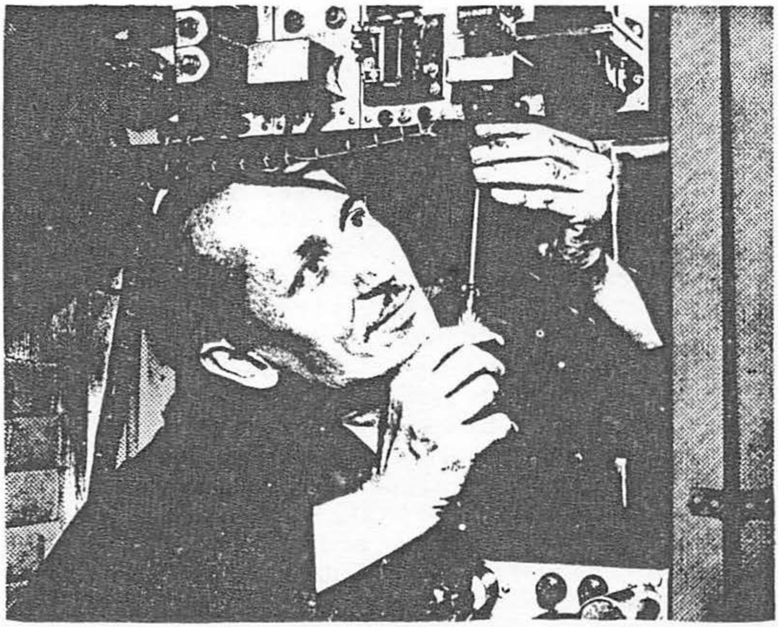


Fig. 1-6. The late John Williams, W2BFD, one of the amateur RTTY pioneers.

frequency shift by means of a small potentiometer. On the VH-F bands we normally use audio frequency shift keying (AF-SK), generating the 2125- and 2975-Hz tone and switching between them or varying an audio oscillator between the two tones. This audio is then fed to the transmitter, and we have a system which is very tolerant of radio-frequency instability. This was of particular value back in the 1940's and 50's when it was difficult to achieve the stability needed for VHF FSK transmission. Most of the early RTTY'ers used surplus SCR-522 transceivers which would receive any RTTY station within 20 to 50 kHz of the channel selected.

AFSK has its place in the scheme of things. A well tuned sideband transmitter can be fed the audio shift signal and it will transmit pure FSK. You can get into trouble if the unwanted sideband is not sufficiently rejected. The FCC sends notes if they catch you with two FSK signals at once.

The Teletype Code

Some sort of signal has to be sent from a keyboard to a typing unit. If you count the number of keys on the keyboard you will see that there are thirty-two (don't forget the space bar). A short consultation with binary arithmetic will show you that we will need a minimum of five pulses to give us 32 characters. You can count it out like this: 2-4-8-16-32.

Each time you press a key on the keyboard you are selecting a specific group of the five units of the code which we will turn into pulses. The letter A is coded 1-2, the letter R is 2-4, etc. If you take a close look at the bottom of a keyboard you will see that the lever attached to key A has notches at positions one and two and no notches at 3-4-5. If you follow the mechanism further you will see that the notches cause levers to operate which lock into place on the distributor. (Fig. 1-7)

The distributor takes the units which are selected by the keys (all are selected simultaneously when you press the key) and spreads this information out into time sequences. The distributor acts as a five-position switch, with each position connected to one of the levers. The motor on the printer turns the distributor at a fixed speed (the same speed as all other printers in the country...you hope). When you press the letter A and levers one and two are locked into place, this releases the distributor for one revolution and sends pulses out during the first two fifths of the cycle. A sixth pulse is added to this in order to synchronize the typing unit distributor or receiving distributor as it is called. Thus both are spinning at the same time and when the transmitting distributor is connected to the first code segment the receiver will be likewise.

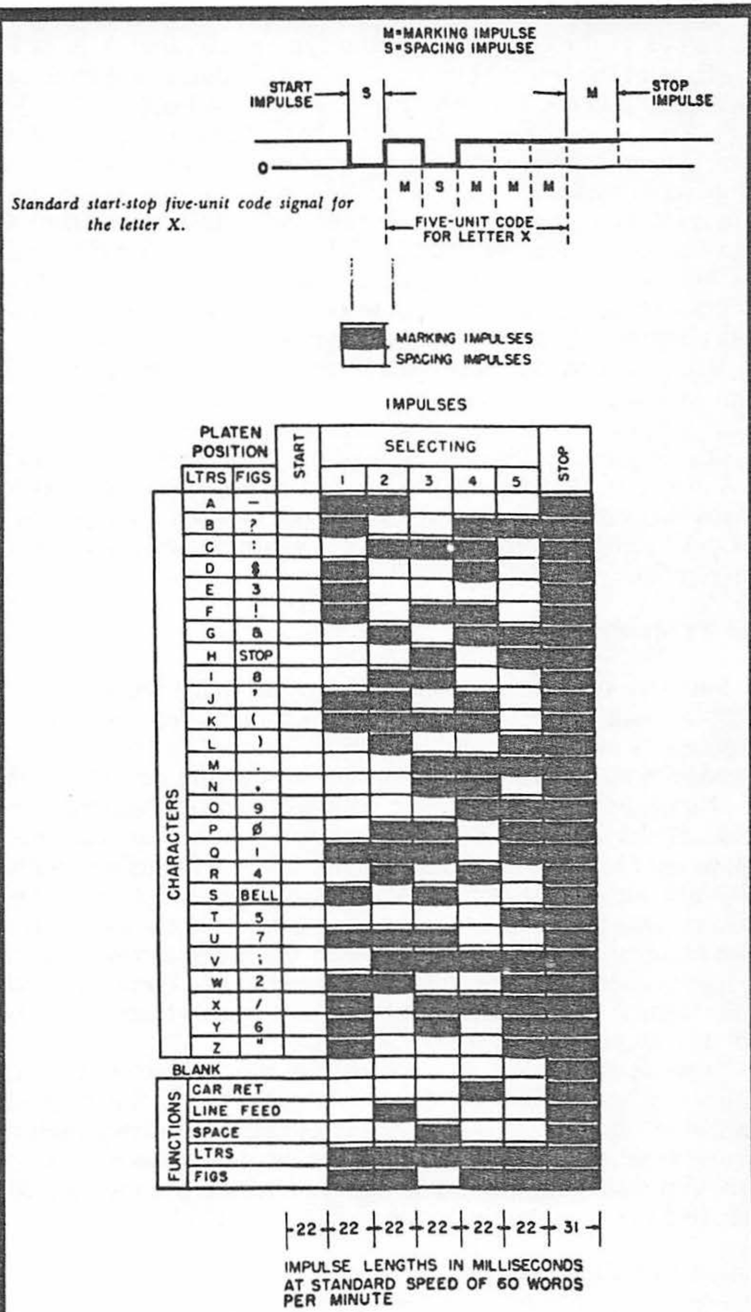


Fig. 1-7. The International Five-unit code.

One more pulse is added to the six we've described; this one comes at the end and tells the typing unit that it is now time to print the selected letter. The first six pulses are each 22 msec long, and the seventh "print" pulse is a little longer, 31 msec. This spare time makes sure that all distributors in the circuit have time to make a complete revolution and come to a halt before the next character can be sent. If you add up all the time involved it comes out to 163 msec per letter, which limits us to about 60 words per minute. This is a pretty good clip and will keep a good typist hopping. Unfortunately, most RTTY'ers are not good typists and the average speed of hand typing is probably about half the capability of the system.

Once we have our pulses all in time sequence we can send them over one pair of wires, or via radio. The first (starting) pulse is always a mark, which means that no current flows; the next five pulses depend on the letter to be sent. In the case of A we are sending space-space-mark-mark-mark, which means that current flows during the first two selection pulses, and not during the last three. The seventh pulse is always a space pulse.

Two Frequencies

Since we use two different frequencies when we transmit RTTY signals over the air, it is convenient for discussion purposes to assign a name to them. These two frequencies correspond to the current flowing or not-flowing conditions of the printer. For standardization we use the upper frequency as the mark channel and the lower is the space channel. This applies only to FSK. We do it just the reverse for AFSK, with the mark signal being 2125 Hz and the space 2975 Hz. The reason is that the station receiver turns the frequencies upside down for us and even though we send the mark as the higher frequency signal, by the time we have changed them back to audio tones it is the lower. You may not care which is which and just want to operate.

Suppose you want to tune up on the 80 meter RTTY channel. What frequency should you use? Well, the official channel is 3620 kHz. If you are going to use a frequency meter to tune your vfo you have to know what goes where. In this case, you would send a mark signal and tune the vfo to 3620 kHz. This would make the space signal 3619.150 kHz.

Tuning in RTTY Signals

When you are first trying out your printer and converter, it is a good idea to test it on one of the ham bands. Though

there are many RTTY signals floating around the commercial bands, there are a lot that sound similar to RTTY and still will not give you any copy. You know the ham signals will print if everything is working.

On the initial tuneup, it is best to turn off the bfo and tune first for maximum signal strength. This makes sure that the signal is in the passband. Then turn on the bfo and tune until the tones are about 2000 and 3000 Hz. A tuning monitor will greatly expedite this process, but it can be done without. As you hit the right frequencies the converter will start to work and the polar relay will start to click in time with the signals. Now, when you turn on the printer, it should print. If it gives out with gibberish you probably have the bfo tuned to the wrong side of the channel. You can either retune to the other side or, if you have a reversing switching on your converter, flick it. This should result in recognizable words. You still may have some difficulty with noise from the printer, but you will get some copy. From there on you can refine the system.

Tuning Indicators

It is fairly difficult to tune in an RTTY signal just by ear. Even after years of practice few amateurs can tell 2125 Hz when they hear it. The best way to know when the receiver is tuned right is to put an indicator of some sort near the receiver which will tell you how you are doing. The earliest system was two 6E5 magic-eye tubes connected to show maximum output of the two tuned audio amplifiers. This is still an excellent system. A scope tube is a little more complex to install, but is perhaps a bit more impressive. In this case, you connect the tuned amplifier outputs to the deflection plates. This indicates the mark signal as a horizontal line and the space signal as a vertical line. You then tune the receiver until the two crossed lines are equal in length and turn on the printer.

RTTY Converters

There are two fundamentally different types of receiving converters. One acts like an FM discriminator and operates at the receiver intermediate frequency. The complexity of these converters has kept them from gaining much acceptance in ham circles. Also they have little to offer over the simpler audio types. The audio converter or terminal unit (TU) is designed to separate the standard tones of 2125 and 2975 Hz and cause a polar relay to be operated or a flip-flop circuit to be actuated.

Let's take the converter step-by-step and see what it is all about. First we have to set about separating the two audio tones. This is done by feeding the audio signal from the receiver into two separate audio amplifiers, both of which have sharply tuned circuits which allow only our desired frequencies to pass. The output of these tuned amplifiers is then rectified to DC. Thus, when a signal is fed to the two amplifiers, the output from each will be about equal until either 2125 or 2975 Hz is fed in, at which time one selective amplifier will have a high output and the other very little. This gives a high DC voltage from one and a small one from the other. Next we amplify that DC voltage so it can operate a relay or key printer magnets.

Refinements can then be added such as a bandpass filter on the input to cut down everything below 2000 Hz and everything above 3000 Hz. This keeps very loud tones other than the RTTY signals from confusing your converter. Without the input bandpass filter you can get some difficulty from the strong heterodynes.

Many tuned circuits have been evolved for the selective amplifiers. These started out back in 1947 with modified AC-DC speaker output transformers, then the FL-8 and FL-5 surplus units became popular, then TV width coils. Now most converters use the 88-mh toroid coils which are available on the surplus market.

Those of you who are in the process of building your RTTY station from scratch would probably do best to use the newest design. This has the advantages of using an inexpensive printed circuit board, using compact transistors, and the circuitry utilizes everything that has been learned from past experiences with other circuits.

CHAPTER 2

Basic Principles of RTTY

The number of radio amateurs using teleprinters on the air is increasing at a great rate. Each of these new amateurs is faced with many technical problems which must be solved. Some are fortunate enough to live near an already established amateur teleprinter station. For those not so fortunate, there has been a great need for down-to-earth information on the manner in which one goes about entering the ranks of the current operators.

TYPES OF MACHINES

There are many different types of teleprinter machines, manufactured by different companies. Many of the ones available to hams have been manufactured by the Teletype Corporation. For example, the models 12, 15, and 19 (Fig. 2-1) are frequently found in amateur circles. Many people who

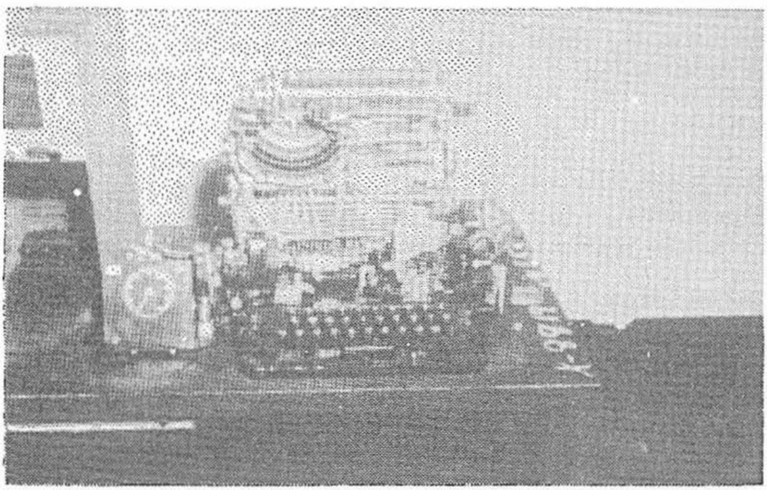


Fig. 2-1. Model 19, shown here with covers removed.

enter into RTTY select the model 19 because it offers flexibility at a reasonable cost. The flexibility results from the tape equipment which is an integral part of the model 19 while the cost is moderate because of the machine's age. (One can purchase a model 19, pre-tested and shipped to the door, for approximately $150.00.) Other models, such as the Model 15, which does not include tape equipment, can be acquired for considerably less. On the other hand, for those who can afford the finest, Model 28 machines are available.

TAPE EQUIPMENT

When we speak of tape we are referring to the 5-channel paper tape that can be used to store information for later use. Information can be placed into paper tape using a paper tape punch. The information is then saved until it is read by a paper tape reader. The information contained on the tape can be alphabetic, numeric, and-or control. By control information, we mean information capable of performing such functions as ringing a signal bell, advancing the paper in the typer, or returning the typing unit to begin a new line. All of this information is stored on the tape in the form of punched holes which by their position and number indicate the particular character of interest.

Tape equipment is popular in the RTTY field, because it enables one to prepare messages ahead of time and to send them automatically when desired. This is accomplished by punching the messages into paper tape and then later reading them with a paper tape reader at a high speed (approximately 60 wpm). Many amateurs enjoy calling CQ by using tape equipment to send out the signals which, when received, generate the normal type of CO format, i.e., CQ CQ CQ DE WAOOBJ K K K. The tape equipment thus represents a way of transmitting from a teleprinter machine without actually doing the typing at the time of transmission.

The Model 19

The Model 19 can be thought of as having six essential parts:

1. A Model 15 page printer
2. A keyboard capable of transmission to an outside circuit, a paper tape punch, or both simultaneously
3. A paper tape punch
4. A paper tape reader and transmitter

5. A sturdy table containing power distribution connectors
6. A suitable DC power supply

The above six parts, when properly connected, enable the operator to perform the following functions:

1. Receive incoming messages on the page printer
2. Prepare paper tapes while receiving incoming messages
3. Prepare paper tapes while simultaneously transmitting directly from the keyboard
4. Transmit from the keyboard to external circuits (while transmitting, the information can be printed on the printer or can be transmitted "blind")
5. Transmit from the paper tape reader
6. Receive on the printer while transmitting with either the keyboard or tape transmitter.

Function 6 is especially interesting for it enables one to operate in a simulcast mode.

HOW THE MACHINES WORK

The teleprinter is activated by releasing the printing mechanism. This is accomplished by deenergizing the selector magnets. In the Model 19, the selector magnets are located on the left-hand side of the machine. With the printer motor running, the printer unit is maintained inactive as long as the printer magnets are energized. When current ceases to flow, the magnets release. If the current is interrupted according to the teletype code, the printer will be activated and the appropriate character will be printed.

The selector magnets can be wired in series or parallel. In general, they are wired in parallel. The magnets are energized with direct current—60 ma being required for proper operation when wired in parallel. The Model 19 is generally supplied with an appropriate power supply which will furnish the current necessary for the selector magnets and also the tape punch and tape reader. The supply is in general capable of supplying 120 volts at 800 ma (1200 ma intermittent) and may carry a Western Electric identification of KS-5928. The resistance of the selector magnets is low and a series current limiting resistor is required when using the 120-volt supply. Do not attempt to operate the unit without the series resistor. The magnets will overheat in a few minutes time and will burn out shortly thereafter. A variable resistor is desirable so that the current can be adjusted when the unit is periodically serviced. A suitable resistor would be a 2500-ohm unit rated at 20 watts.

The selector magnets do not have to be operated with a high voltage DC supply. Low voltage supplies will work quite well and reduce the danger associated with the higher voltages. The main advantage of using high voltage is the self-cleaning of contacts which takes place due to the slight arcing.

A Local Loop

In teleprinter installations, a closed circuit is referred to as a loop. Thus a local loop is a closed circuit which encompasses equipment in your local area, i.e., only your machine. A local loop could consist of a printer and a keyboard or a printer and a paper tape reader. To set up a local loop with your Model 19, you will have to locate the leads connecting to the selector magnets, the keyboard, and the paper tape reader-transmitter. Details on how to locate these leads will be presented later in our discussion.

The inter-wiring required for successful operation can be greatly simplified by thinking of the units performing like the following components:

Printer: An electromagnet which is normally energized.
Keyboard: A SPST switch which is normally closed.
Tape Reader-Transmitter: A SPST switch which is normally closed.

With the above understanding of the units, we can immediately wire a local loop containing the keyboard and printer. The loop would be a series circuit containing the following elements:

1. The keyboard SPST switch
2. The printer's electromagnet
3. A current control resistor
4. A source of direct current

Such a circuit is shown in Fig. 2-2. Since the keyboard switch is normally closed, a current will flow through the selector magnets keeping them energized with an amount of current governed by the setting of the current control resistor. When a key on the keyboard is depressed, the normally closed contacts will be interrupted momentarily. This will cause the selector magnet to de-energize and thus activate the printing mechanism. Note that the circuit is not simply opened and then closed again. The actual interruption may consist of several openings and closings—the number and spacing depending upon the character being sent by the keyboard.

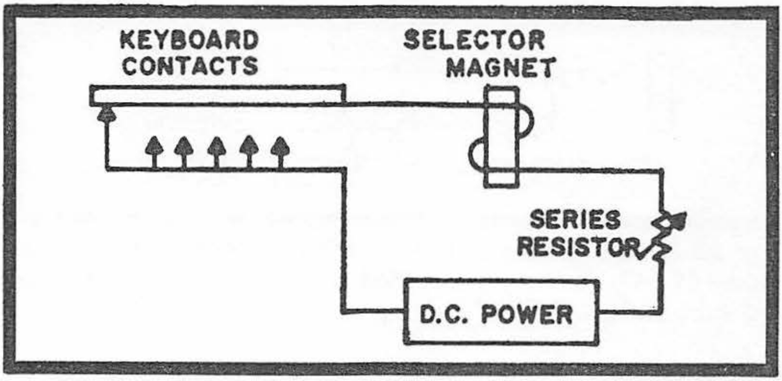


Fig. 2-2. With the local loop shown here, the Teletype operator can type on the keyboard and activate the printer for printing.

With this simple local loop, one can type on the keyboard and have the characters displayed on the printed page by the printer. Now observe that the tape transmitter is simply another normally closed switch. If the above loop is opened and the tape transmitter inserted, we would have a new loop in which the printer could receive from either the keyboard or the tape transmitter. Just as depressing a key on the keyboard causes the circuit to be interrupted momentarily, so passing tape through the tape reader causes the same thing to happen. Such a loop is shown in Fig. 2-3.

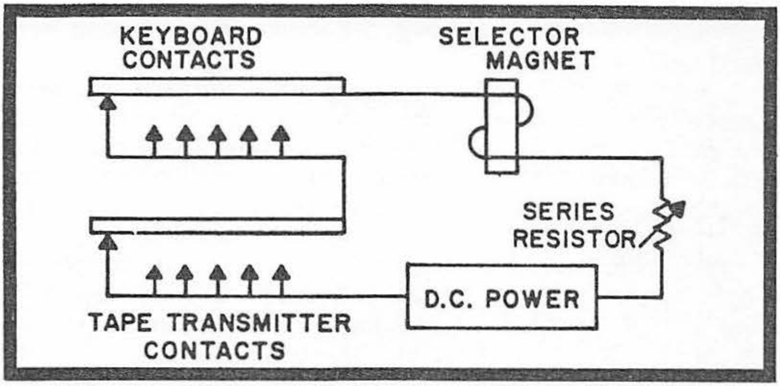


Fig. 2-3. With this local loop arrangement the printer will print from either the keyboard or the tape reader.

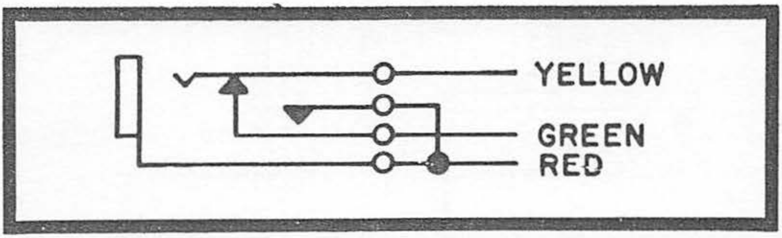


Fig. 2-4. Color coding of the switching jacks found in the Model 19. With the plug inserted, connections are made to red and yellow.

Unit Wiring

Although you will find publications showing a wiring diagram for the Model 19, this does not necessarily mean that your Model 19 will be wired exactly as shown. At first this sounds as though it would pose great problems. However, a few minutes spent exploring the machine with an ohmmeter will pretty well identify the necessary wires.

In general, you will find two wires coming from the printer which are terminated with phone plugs. One will be red and the other will be black. It is these wires which you will want to examine with an ohmmeter. Sometimes one is the keyboard while the other is the selector magnet. Other times, the two centers of the plugs are the one unit while the two outsides are the other. You will have to study your machine to determine exactly how it is connected.

Switching jacks are provided for the phone plugs. These jacks are mounted in the Model 19's table along with the various power connectors. Fig. 2-4 shows how the jacks are wired and color-coded.

Assuming that you are using the standard Western Electric KS-5928 power supply, the power connections are straight-forward. On the back of the table, you will find various sockets that will accept only special keyed plugs. These plugs are contained on either the power supply unit or the printer.

The power from the incoming line is supplied to the power supply by the toggle switch located under the table top (left hand side). This switch represents the main power switch for the entire teleprinter machine. The printer's motor is controlled by the on-off switch located on the front of the printer. Some units also have a switch to control the AC to the tape reader's motor. If your machines do not have such a switch one should be installed.

Provision is made for mounting a switch in the box containing the general power switch mentioned above. If your Model 19 does not have the switch for controlling the tape reader's motor, you will find that one wire comes into the switch box, loops around, and then goes back out. The new switch should be inserted in this loop.

The on-off switch mounted on the front of the tape reader-transmitter is not a power switch controlling the AC drive motor. The switch is connected in series with a solenoid used to maintain the transmitter inactive. If you lift off the cover of the transmitter, you will find a large solenoid located near the front, on the left hand side of the unit. If this solenoid is energized, the motor will turn the vertical shaft and "read and transmit" the information contained on paper tape. Thus to send from tape, two sources of power must be connected to the tape reader—the AC power to the motor and the DC to the control solenoid. The DC for this solenoid is distributed via the printer's inter-unit switches (see next section).

Inter-Unit Operation

There are several modes of inter-unit operation possible with the Model 19. We indicated before that one must identify the leads coming from the keyboard and selector magnet. We implied that they were directly connected. Actually, there are switches installed in these lines. You will find three switches on the front of the printer. The one on the far right is the "on-off" switch for the printer motor. The motor must be turned on to either receive with the printer or to transmit with the keyboard. It does not have to be on to punch tape (blind punching). Now observe the switch on the left hand side of the printer. This switch is connected in series and in parallel with the keyboard. In the "send" position the keyboard is connected to the out-going wires. In the receive position, the keyboard is shorted (bypassed). In the "break" position the keyboard circuit is opened. The remaining switch, located slightly left of the printer motor power switch, selects the mode of transmission that will take place from the keyboard. Thus, if in the KEYBOARD position, transmission will take place from the keyboard to the outgoing lines. In the KEYBOARD AND TAPE position, transmission goes to the outgoing line and the information being typed is simultaneously punched into paper tape. When placed in the TAPE positon, the keyboard will punch tape but will not transmit to the out-going line. In this mode of operation, the printer motor need not be running.

We should now note some interactions which take place between the paper tape reader-transmitter and the above mentioned switches. If you want to send something, whether it be from tape or the keyboard, the send-rec-break switch must be in the SEND position. Furthermore, if you want to send from paper tape, the keyboard-kbd & tape-tape switch must be in the KBD TAPE or TAPE position. Also, when transmitting tape, remember to turn the solenoid switch to on. Note: Some tape transmitters have other switches connected in series with the solenoid switch. These include switches to stop transmission if there is no tape in the reader, etc.

OPERATING

Up until now, we have concerned ourselves with the machines used to send and receive teletype. But we have not talked about applications involving anything other than a local loop. Certainly the fascination of typing on an over-sized typewriter will wear off quickly. Let us then pursue the reception of teletype signals and the manner in which they can be used to activate a teleprinter machine.

You will recall that we said the printer was activated by interrupting the flow of current through the selector magnets. It then follows that if we could get the receiver to interrupt the flow of loop current in step with the intelligence being received, we would have a setup capable of receiving teletype signals and converting them into the printed word. If the teletype signals were transmitted in a make-and-break fashion, we could, for example, rectify the receiver audio output and apply it to a normally open relay (so long as the signal was present, the relay would be energized and the contact would be kept closed). Thus when the audio ceased, the relay would open the loop to the printer and de-energize the selector magnets and printing would take place. Such a circuit, although lacking many desirable characteristics, can and indeed is used by some amateurs.

Modes of Operation

In general, two types of teletype transmission are employed by amateurs: frequency shift keying (FSK) and audio frequency shift keying (AFSK). Frequency shift keying is accomplished by adding and subtracting a small amount of capacitance at the cathode of the transmitter's vfo. The addition and subtraction are done in step with the conditions existing in the local loop. The addition or subtraction of

capacitance to the cathode of the vfo causes it to shift its frequency of oscillation. Thus, if current is flowing through the selector magnets, one frequency is transmitted. If current is not flowing, the alternate frequency is transmitted. Thus the condition existing in the loop is constantly indicated by the frequency being transmitted. In audio shift keying, the same theory applies except two audio notes are transmitted to indicate the conditions existing in the loop. This is accomplished by switching the transmitter's audio input from one audio oscillator's output to that of another.

The devices used to convert the received teletype signal into pulses capable of operating the printer magnets are called converters or terminal units. There are many different unit circuits available and the literature abounds with information about them. Some converters process the audio output of the receiver while others process the signal existing in the IF section of the receiver. The output of the converter amounts to a switch which is either open or closed. As soon as you recognize this fact, you should realize that the converter is very similar to the keyboard or tape transmitter so far as its function as a circuit element. Since the converter output is just another SPST switch, we can set up another series loop containing the converter's switch, the selector magnets, the series limiting resistor, and a source of direct current. (See Fig. 2-5.) As the receiver's output changes in step with the intelligence being transmitted, the converter's switch will

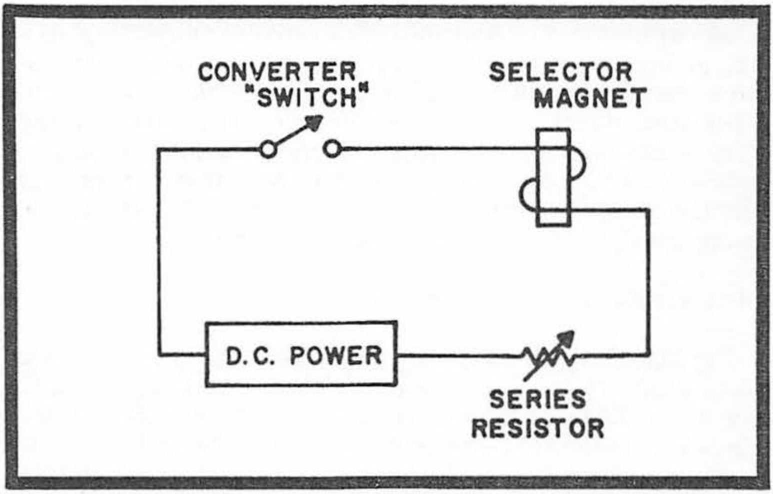


Fig. 2-5. Installing of the RTTY converter in the local loop to energize the printer from received RTTY signals.

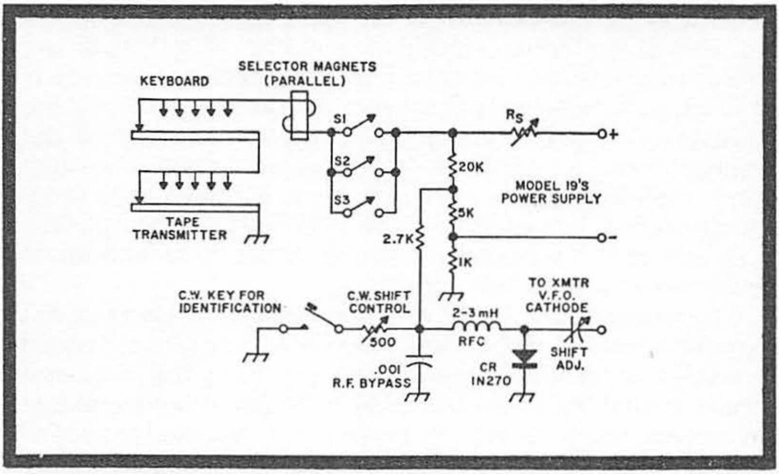


Fig. 2-6. The Mainliner frequency shift keying circuit. The resistor RS is adjusted for 60 ma through the selector magnets.

open and close and thus set up current changes in the receiving loop identical to those existing at the transmitting station.

To keep the converter from responding to signals other than the one desired, it is equipped with various filters which are designed to reject the unwanted frequencies.

To summarize, transmitting involves converting the make-and-break in the local loop into either a shift in transmitter carrier frequency or a change in the audio note transmitted. Receiving involves converting the change in the audio frequency received (audio type converters) into a make-and-break signal for the receiving loop. (Note that in receiving FSK, the receiver's bfo is turned on and beat with the incoming signal just as in receiving CW signals.)

Shift Circuits for FSK Operation

Various shift circuits are available. One very flexible circuit is that called the Main-liner. This circuit uses a small diode in a diode switching circuit to add or subtract the capacitance to the vfo cathode. The circuit is shown in Fig. 2-6. The Model 19's power supply can be used to power the circuit.

The circuit's operation centers around the voltage drops appearing across the various resistors. With the loop closed, a voltage of a given polarity exists across the diode. Opening the

loop (sending a character with the keyboard or tape) causes the polarity across the diode to reverse. As the polarity changes, the diode either conducts or does not conduct. Thus the diode either shorts the capacitance applied to the cathode to ground (thus connecting it into the circuit) or disconnects the capacitance by allowing the one lead to float.

Station Setups

Every amateur teletype station is set up differently. A typical setup is shown in Fig. 2-7. Once the basic principles are understood, one can design any configuration desired. Although teleprinter machines may vary in detail, the basic principles are all the same. An understanding of the manner in which the machines operate, coupled with the electrical competence required for the general class license, should enable the interested amateur to install and successfully operate an amateur radio teletype station.

FROM TELEGRAPHY TO RTTY

The first electrical communications technique was telegraphy. Samuel F. B. Morse is generally credited with the invention of telegraphy, although, like Marconi in the case of radio, Morse's major contribution was to gather a number of discoveries made by others into a working system and

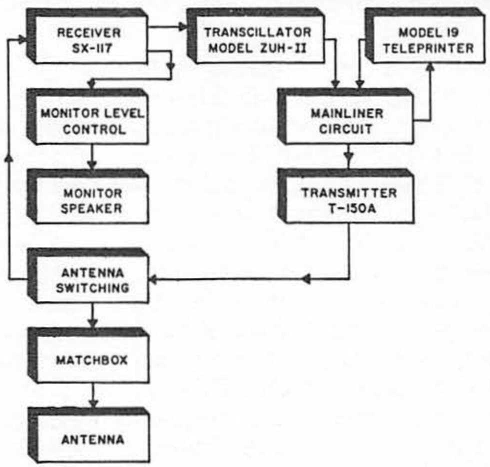


Fig. 2-7. A station setup.

promote it to the world. The familiar hand key was, for several decades, the only means of originating any electrical communication.

As the years passed, a number of inventors attacked the problems imposed by the hand key and its requirement for an operator who knew the telegraphic code. Eventually, Joy Morton and Charles L. Krum developed a machine which was able to bypass the operator requirements. It was, essentially, a cross between a typewriter and a telegraph system. In 1915, the Associated Press adopted the Morkrum machine.

Meanwhile, Edward Kleinschmidt independently came up with a machine to perform the same purpose. In 1925, the Morkrum and Kleinschmidt companies merged, and five years later AT&T bought the firm. The machine was named Teletype by blending the prefixes of telegraph and typewriter, and the firm was named the Teletype Corporation. The word "Teletype" is, incidentally, still a registered trademark of the Teletype Corporation and cannot legally be used to refer to any other make of teleprinter equipment. Its common abbreviation, TTY, is not a trademark and can be used as a general term.

TTY machines were designed with the requirements of land-line telegraphy in mind, but during World War II it became apparent that radio transmission could replace the land-lines with little change in techniques—and RTTY (radio TTY) was born.

Growth of Amateur RTTY

Amateur RTTY got its start in 1946 when a number of hams on both coasts, in a parallel but apparently unconnected action, persuaded several companies to release obsolete machines to ham use instead of smashing them for junk.

Today, TTY equipment has many uses besides the original land-line telegraphic application, and as a result machines are not too difficult to locate. For instance, TTY equipment is widely used in the digital computer industry (where the machine is known as a remote terminal) and a number of firms started producing machines for this purpose which had never participated in the telegraph-communications use.

The principle upon which TTY operates is similar to that of normal CW transmission, with one major difference. Where CW makes use of five different kinds of signals—dits, dahs, and three lengths of spaces to indicate in-the-character space, word space, and sentence space—TTY uses only two. The two kinds of signals used by TTY are called "mark" and "space"

and may be represented in several different ways. The advantage gained by using only two kinds of signals is that a machine can then interpret the signal with a minimum of logic or decision-making circuits; conventional CW with its one-unit dit, three-unit dah, and one-, three-, and five-unit spaces would require much more interpretation to decode. This is simple for humans, but most complicated for a machine to perform (although machines have been built to do so).

Several Codes

Because of the differences in the kinds of signals used, TTY equipment does not employ the International Morse Code. Instead, it uses one of several other codes which assign characters to combinations of the mark and space signals. Amateur and most commercial equipment uses the International Five-Unit Code, but equipment designed for use with computers is more likely to employ the American Standard Code for Interchange of Information or a different code originally introduced by the Friden Company for its Flexowriter equipment.

Use of Codes

Regardless of the code used, the signals sent and received by a TTY machine consist of groupings of mark and space signals. At the machine itself, mark is usually represented by the presence of current in the line, and space by the absence of current, so that they can be thought of as on and off conditions. This same representation can be used in the circuit from one machine to another, and the result is known as make and break keying. Ordinary CW is sent by make and break keying, for instance. For the machines, however, the use of make and break keying makes it impossible to distinguish an extended space condition from a circuit failure and resulting total loss of signal—and so other representations are usually employed instead.

The most common technique used to represent mark and space conditions in RTTY is to use two radio frequencies rather than just one. One frequency represents mark and the other is space. Now a loss of signal can be recognized by the absence of both the mark and the space signals, while an extended space condition is signaled by the presence of the space signal.

The two frequencies are usually very close together. The difference between the mark frequency and the space

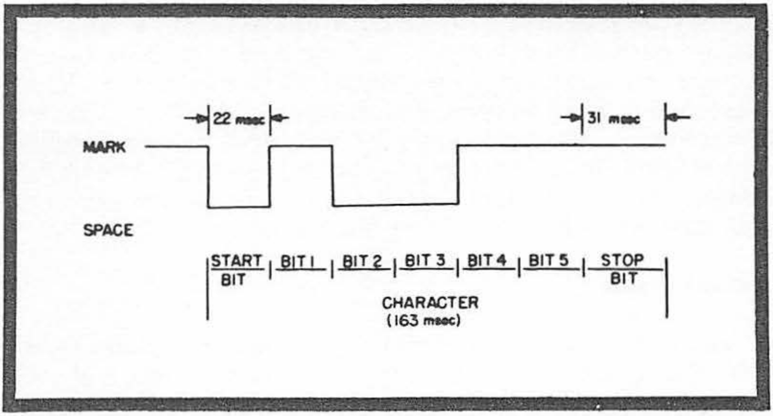


Fig. 2-8. This waveform shows the appearance of the character B as transmitted in the asynchronous serial RTTY code. While code contains only five units, character as sent includes seven. Every character has a space start unit, and a mark stop, unit, in addition to the five units which carry character's meaning. All bits except stop bit are equal length, 22 msec at 60 wpm speed. Stop bit is longer, 31 msec at 60 wpm, to pad out to transmission rate and allow time for equipment to stop between characters.

frequency is known as the shift, and the normal shift is only 850 Hz. This tends to minimize fading because the two signals will fade together. Narrow-shift is also used, with a shift of as little as 150 Hz, to minimize fading effects still more. Any shift less than 900 Hz is legal for use.

It makes no difference which of the two frequencies is used for mark and which for space, but common ham practice is to use the higher of the two frequencies for mark. In case one station chooses to reverse this, all the other station needs do is to reverse the polarity in his receiver—which may be done by merely tuning to the other side of zero beat. The situation is exactly the same as the choice between upper and lower sidebands in SSB operation.

The International Five-Unit Code

In TTY work, again regardless of the actual code used (and for the rest of our discussion we will assume that the International Five-Unit Code is the one to be used, since it is the one required for ham RTTY by FCC regulations), each character begins with a start signal and ends with a stop

signal. This makes the total character length with a five-unit code actually seven units.

The start signal is always a space and the stop is always a mark. These two signals establish timing synchronization between the transmitting machine and the receiving machine, and thus make it possible for the intervening code signals to be decoded.

When no character is being transmitted, a steady mark signal is sent. The space condition produced by the start portion of a character then starts the machine's decoding mechanism. The decoding mechanism sorts out the next five signal conditions to determine what character is being sent, and stops. The stop portion of the character simply provides enough time for the decoder to come to rest before another character can be sent.

Normal speed for amateur RTTY equipment is 60 wpm. At this speed, each part of a single character (except for the stop) lasts for 22 milliseconds. The stop is half again longer, or 31 msec, for a total character duration of 163 msec.

A TTY machine has the appearance of a typewriter, its major parts so far as the operator is concerned are the keyboard and the printer.

Keyboard Action

When a key on the keyboard is pressed, it sets up in some type of memory device the actual code to be transmitted. This memory device may be either mechanical or electronic. The machines in general usage all employ mechanical memory; pressing a key pushes down a notched lever, which then latches in the down position. Each key has a different lever, and the notches on each lever correspond to the space portions of the character for that key. All these notched levers rest on top of and across another set of five levers, so that when a notched lever is latched down it pushes down any of the second group of levers which do not have notches above them, and leaves up any of the second group which are matched by notches.

This second group of levers operates a set of five contacts, one for each unit of the character code. A motor-driven distributor then wipes a selector brush across the five contacts in turn (the start bit of the character is built into the selector mechanism, as it is the stop bit at the end). If the contact lever for any one unit is up, a space is sent when the selector brush wipes across that contact; if the contact level is down, a mark

is sent. When the selector brush leaves the last contact, the distributor motor stops and the keyboard is unlatched in preparation for the next character to be sent.

Printer Action

We have seen how the operator's pressure on any one key is translated into the code pattern to be transmitted to represent that character. The reverse operation, translation of a received code pattern into a character to be printed, is accomplished in much the same way by the printer. However, the printers of the various models of TTY machines are the areas in which the most drastic differences occur.

The explanation of printer action which follows is based on the action of the Teletype Model 12 machine, which is now obsolete. More recent machines operate in substantially different ways, but the Model 12's operation serves to explain the basic idea without complicating the description with the complexities of single-magnet operation.

The received signal is applied to the printer as a make-and-break keyed current, with mark represented by the presence of current and space represented by its absence. In the absence of any received signal, mark condition exists and current is flowing through the printer line.

In the printer, this current flows through a latch relay and six selector magnets. So long as current flows through the latch relay, the printer is inactive.

Distributor Operation

When a character appears, its arrival is indicated by its start bit which is a space condition. This drops out the latch relay and permits the drive motor to turn the receiving distributor through one revolution.

As the receiving distributor rotates, it connects each of the six selector magnets, in turn, to the signal line at 22 msec intervals. Each selector magnet is either tripped or unaffected by the signal line; if the line condition is mark when the selector is connected to the line, the magnet is tripped, and if the line condition is space nothing happens.

The first five selector magnets control notched levers, which, in the tripped position, block movement of the key levers that do the actual printing. When all five of these notched levers have been tripped, only one key lever can match the notches in all of them. Every other key lever will fail to match a notch in at least one of the code levers.

Printer Motor Action

The sixth selector magnet, which is tripped by the stop bit, permits the printer motor to drive the key-lever mechanism and print the one character which matches. After the printing, the entire mechanism is reset to its initial conditions and the receiving distributor stops, ready to start again which the next start bit arrives.

In later machines, only a single selector magnet is used and its mechanical action is routed to the appropriate points for each bit by a cam which sets up toggle action at the proper time. Some machines do not use key levers; instead, they position a type box in front of a hammer so that the selected character will be printed when the hammer strikes.

Non-Printing Characters

Not all characters in a TTY machine cause printing. Some, such as "carriage return," "line feed," "letters," and "figures" cause mechanical action in the printer instead.

Local Printout

In a land-line setup (called a local loop when one is set up in a ham shack, as for test purposes), the keyboard contacts are connected in series with the printer selector magnets and a power supply which frequently is 150 volts DC at 20 or 60 ma. Pressing a key causes generation of the code character, and the serial code character operates the selector magnets and causes the printer to print. If this all happens in the same machine, the result is an electric typewriter; if the keyboard and printer are in different machines, the result is telegraphy without Morse code.

Wireless TTY

To convert the telegraphy setup to radio, all we need do is find some way to put the mark and space information on RF waves rather than on copper wires, and then recover it to drive the printer.

We could simply substitute the keyboard for the hand key in a CW installation, and we would have make-and-break keying. The first amateur work over long distances was done in this manner because no other form of teletype operation was permitted by the FCC. At the receiver, the audio was converted to DC current to drive the selector magnet and that was all it took.

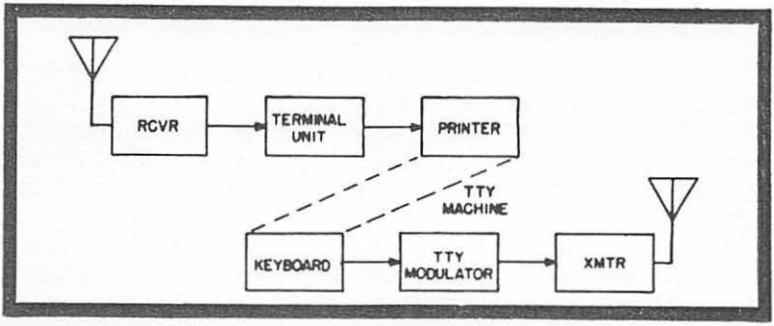


Fig. 2-9. Block diagram of typical RTTY station. AM station would be same except that speaker would replace terminal unit and printer at receiver output, and microphone would replace keyboard. AM or FM modulator or sideband generator would replace TTY modulator at transmitter input.

However, make-and-break is no longer used in RTTY. Frequency shift keying (FSK) is almost universally used. The frequency shifted may be the carrier itself (FSK), or an audio frequency modulated upon the carrier (AFSK), FSK is used in the HF bands, but AFSK is used by VHF RTTY because of the difficulty in controlling small shifts at high frequency.

The Basic RTTY Station

A typical RTTY station is shown in block diagram form in Fig. 2-9, and compared to an AM station. You can see that the major differences lie in the TTY machine itself, and the circuits connecting it to the transmitter and the receiver.

The transmitter and receiver serve the same purposes as in any other radio communications system. The TTY machine's keyboard generates the proper code for the character to be sent, and the TTY modulator converts this code into the proper modulation for the transmitter (frequency shift or FM for FSK, and a shifting audio frequency plus audio modulator circuits for AFSK). The converter or terminal unit converts the receiver's output into a make-and-break-keyed current suitable for the TTY machine's printer, and the printer then converts this intermittent current flow into a character image on paper.

Station Identification

In the early days, amateur RTTY stations were required to identify themselves by ordinary CW at 10-minute intervals,

as well as identifying in the TTY transmission. Rules have since been modified to permit all identification to be done in TTY. The general requirements for identification are the same as for all other amateur stations; both the transmitting station and the receiving station must be identified by call sign.

TRANSMITTING AND RECEIVING RTTY

We have already examined the means by which the operator's pressure on any one key of the keyboard is converted into an electrical code representing the chosen character, and the received electrical code is converted back into a printed representation of the character. But what goes between?

Choice of Modes

Of course, a radio transmitter and a receiver are used, but the TTY modulator and the converter or terminal unit are the items we are most interested in at this point. There are almost as many different TTY modulator circuits and terminal unit designs as there are RTTY enthusiasts, but most of these different designs have many items in common. The major differences, in fact, are brought about by the choice of FSK or AFSK for the modulating technique, and by the choice between FM discrimination and simulated make-and-break for the terminal unit.

If FSK is to be used, the TTY modulator must vary the frequency of the transmitter, and the terminal unit may either generate the driving signal to the printer directly from the receiver's IF, or indirectly from audio tones produced by the bfo.

If AFSK is to be used, the TTY modulator must first provide a frequency-shifting audio tone and then modulate the transmitter with this tone. Except for the keyed audio oscillator, normal AM transmitter techniques are usually used. The terminal unit for FSK must be of the audio variety.

Since an audio terminal unit can be employed for either AFSK or FSK operation, most of the popular designs have been based on this principle. For FSK-only use, however, the straight FM technique can provide the ability to dig much deeper into the interference of a crowded band for solid copy of a weak signal.

Let us examine at least one of each of these types of equipment to see how it functions. Since RTTY in amateur

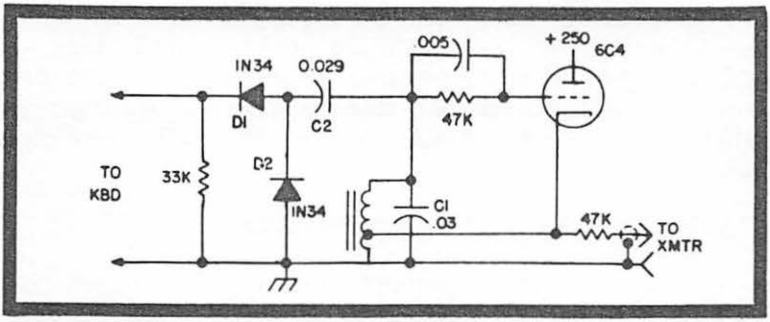


Fig. 2-10. AFSK oscillator. W2PAT inductor is TV width coil. Output of this oscillator feeds conventional AM transmitter's mike jack.

radio began with AFSK, we will examine an AFSK modulator and terminal unit first, followed by the FSK equipment.

An AFSK TU

One of the most popular RTTY terminal units ever described was that designed by Marvin Berstein, W2PAT. This unit provided an introduction to RTTY for a majority of the pioneer RTTY enthusiasts and more to our point, is an excellent example of a complete AFSK unit. It includes both the oscillator to provide a keyed audio tone to a conventional AM transmitter, and the receiving converter to turn received tones into series character codes to operate the printer.

The Oscillator

The AFSK oscillator (Fig. 2-10) is almost completely conventional. The Hartley circuit provides a stable sine wave output with a minimum of components, but any other oscillator circuit could have been used (and most other circuits were, in one or another of the various designs which followed). The part which makes this circuit different from an ordinary oscillator is the diode switch and frequency-shift capacitor circuit, consisting of resistor R1, diodes D1 and D2, and capacitor C2.

AFSK operators normally indicate mark condition with a tone of 2125 Hz, and space with a tone of 2975 Hz. The oscillator is tuned, by adjustment of the coil inductance and the value of capacitor C1, to produce output at 2975 Hz when the keying circuit is open-circuited.

When the keying circuit is closed, R1 is shorted out. This permits diodes D1 and D2 to switch capacitor C2 into the circuit in parallel with C1, thus lowering the output frequency.

The value of C2 is adjusted by trial and error until the output frequency is 2125 Hz.

Diode Switches

Operation of this diode switch may appear a trifle mysterious at first, since no external power is supplied to turn the diodes either on or off. What actually happens is that the diodes rob enough power from the oscillator tank circuit to develop their own switching voltages. D2 conducts for half of each cycle regardless of the keyboard contact position; during the half-cycle that D2 is off, D1 attempts to conduct. The current flow through D1, however, must flow through R1, and a voltage is developed at the "hot" end of R1 which tends to keep C2 isolated from ground.

When the keyboard contacts are closed, R1 is no longer in the circuit and the current flow is direct to ground on both halves of the audio cycle (through D2 on one half and D1 on the other). The diodes now act just like a closed switch, and cut C2 into the circuit.

Because of the diodes, the keyboard contacts handle only a small amount of DC rather than being required to switch AC or high-current DC. This minimizes RF interference to the receiver. In RTTY, this is an important point, because the printing action on the machine normally results from the signals which have gone all the way through the system, rather than on any direct connection between keyboard and printer.

Receiving Converter

The receiving-converter portion of the terminal unit (Fig. 2-11) is not so familiar-looking as the oscillator. It consists of a double-diode limiter followed by a two-stage amplifier which is operated in an overdriven condition to serve an additional limiting function. Output of the active limiter is then applied to the detector circuit, and detector output drives the final current amplifier. The current amplifier operates a relay, and the relay contacts drive the printer selector magnets.

Detectors

The active limiter provides an output level of 15 volts, plus or minus 1 db, with input signals varying over the range of 450 mv to more than 30v. This 15 volt signal is applied to the detector, which actually consists of two separate grid-leak detectors, each tuned to a separate audio frequency. One half

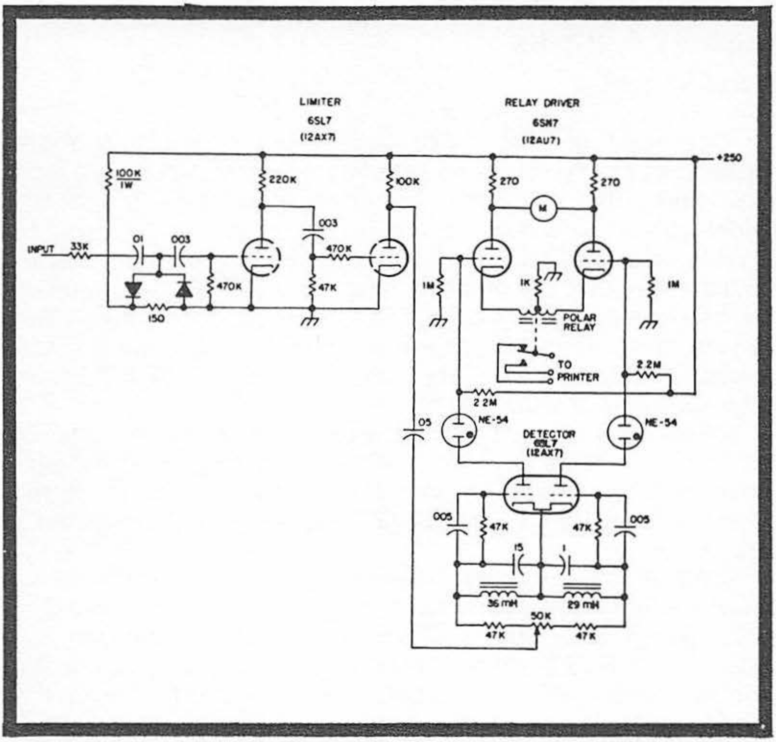


Fig. 2-11. Receiving-converter (portion of W2PAT design) less power supply. Neon tubes at lower center of diagram are type NE-54. Inductors in filters are TV width coils.

of the circuit is tuned to the mark frequency and the other half to the space frequency. Since only one of these two frequencies is present at any one time, one side of the circuit is always drawing more current than is the other side, and this swing of the heavy current flow from one side to the other in the detector stage corresponds to the mark-or-space condition of the TTY signal.

The plates of the two detector circuits are coupled through neon bulbs to the grids of another dual-triode tube. The neon bulbs act as threshold devices. Until firing voltage for the neon is reached, grid voltage on the final tube remains zero. When current flow in one side of the detector drops due to lack of signal at the associated frequency, the plate voltage rises, and when the neon fires the grid of that side of the current amplifier tube goes positive. This makes that half of the final tube draw heavy current, thus pulling the polar relay to the corresponding position.

When the AFSK signal's frequency returns to that to which the detector is tuned, the detector stage current increases and plate voltage drops. When plate voltage falls below about 55 volts, the neon lamp goes out and the grid of that side of the current amplifier returns to ground voltage. At the same time, plate voltage in the other half of the detector was rising and turning on the other side of the current amplifier, thus reversing current through the output relay and driving it to its other position.

Tuning Indicator

The milliammeter connected between the plates of the current amplifier stage is a tuning indicator. When tuned to a TTY signal which has a mark-to-space ratio of 1:1 (as much time spent in mark condition as in space condition), the meter will indicate zero average current. This is because the meter movement is unable to follow the rapid fluctuations of current, and if the mark-to-space ratio is 1:1, the average current fluctuation will be zero.

Test Signal

Another way to provide such a test signal is simply to key an AFSK oscillator with a square wave or an electronic key set for all dits: the RYRYRY sequence is traditional, but the keyed oscillator is more reliable since it does not depend upon the keyboard being in proper adjustment.

Other AFSK TU's

Many other AFSK units have been described and put into use since the W2PAT unit made its appearance; a large portion of those in use today use transistors rather than tubes. The basic principles of all are similar, however, in that the two audio tones are split into separate channels and the TTY code is recovered by detection of both channels. The limiter is also a usual feature since it provides the ability to operate with a minimum of critical receiver adjustment between stations.

An FSK Setup

FSK equipment is similar in many ways to AFSK gear, but differs in some important respects. The modulator consists of a circuit to switch capacitance across the tuning circuit of the regular transmitter vfo (crystal oscillators are difficult to get enough shift on). Such a circuit is shown in Fig.

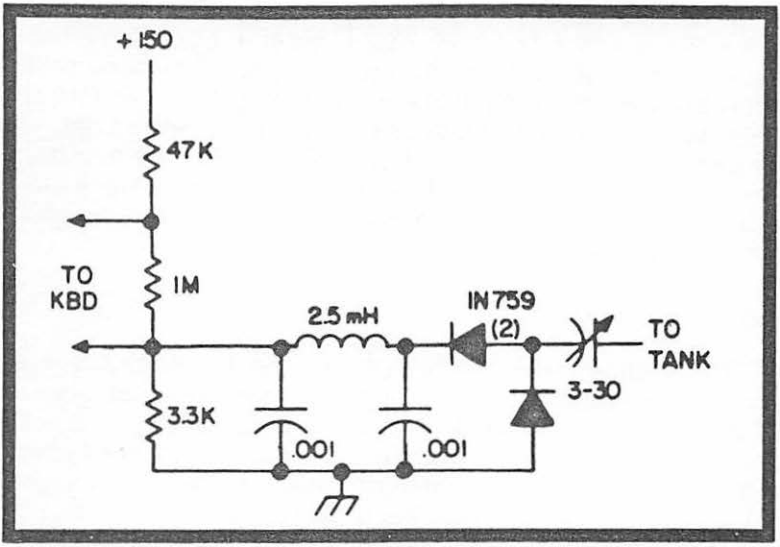


Fig. 2-12. Typical switching circuit for direct FSK. 3 to 30 pf trimmer is coarse adjustment on maximum amount of shift attainable; meg pot is operating adjustment. All components to right of RF choke should be mounted as close as possible to vfo tank circuit, and be made mechanically solid to avoid frequency instability. Components to left of RF choke may be located anywhere. Keyboard contacts must be insulated from ground for use in this circuit.

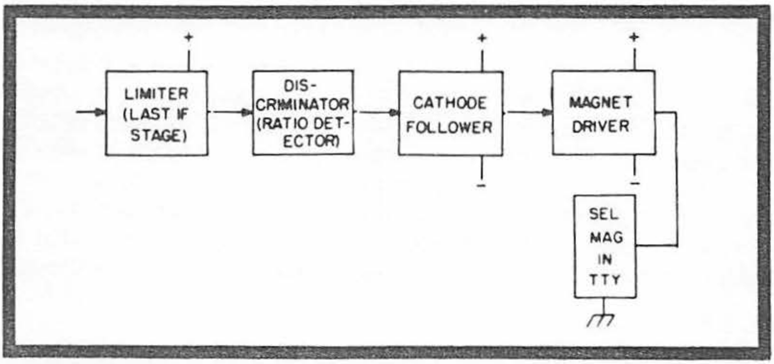


Fig. 2-13. Block diagram of typical direct-FSK or IF RTTY terminal unit. Limiter and FM detector are completely conventional, but DC coupling is employed from FM detector output through to TTY printer selector magnets.

2-12. Like the AFSK oscillator, a diode switch is used, but this diode switch is powered by voltage stolen from the transmitter B+ supply. The function of the switch is to connect the added capacitor C1 across the tuning capacitor when the keyboard contacts are open, and to remove it when the contacts are closed. This makes mark the lower frequency and space the higher one. Note that this is the exact reverse of AFSK practice.

Frequency Shifting

The frequency shift is usually 850 Hz, but any shift less than 900 Hz can be used legally. The variable resistor controls the amount of shift, by varying the effect of the added capacitor. When changing from one band to another, this resistor normally must be adjusted to take into account the frequency multiplication introduced by most transmitters at the higher bands.

Many RTTY enthusiasts receive FSK by using AFSK terminal units, converting the FSK signal to AFSK by turning on the receiver bfo and tuning carefully.

FM Detector

However, an FSK signal is an FM signal, and an FM detector operating at the receiver's intermediate frequency can in many cases give superior performance. The FM detector can be any of the conventional FM detector circuits: discriminator, ratio detector, or pulse counter. Its major point of difference from audio AM detectors is that it must respond to low-frequency signals, since the 2.2 msec duration of a TTY bit is about 23 Hz in sine-wave terms.

Such a direct-FSK terminal unit is shown in block-diagram form in Fig. 2-13. Both the limiter and discriminator stages are completely conventional. The cathode follower between the discriminator and the driver stage permits direct coupling for good low-frequency response.

The driver stage is a switching circuit, so hooked up that a mark signal produces current flow through the output terminals and a space signal stops the flow of current. Both positive and negative power supplies are used, so that the output terminals will be at approximately ground potential to minimize shock hazards.

CHAPTER 3

Equipment

To provide a closer look at some of the classic machines and associated circuitry, here are details on the Model 14 typing reperforator (Figs. 3-1 and 3-2), the Model 15 (Figs. 3-3 through Fig. 3-5) and other equipment, including some designed for Armed Forces use and frequently available as surplus. Some of these machines have already been described and pictured as examples in earlier chapters.

MODEL 19

The Model 19 machine is similar to the Model 15, with several exceptions. The keyboard on the Model 19 has a tape perforator on the left side. This piece of equipment makes it

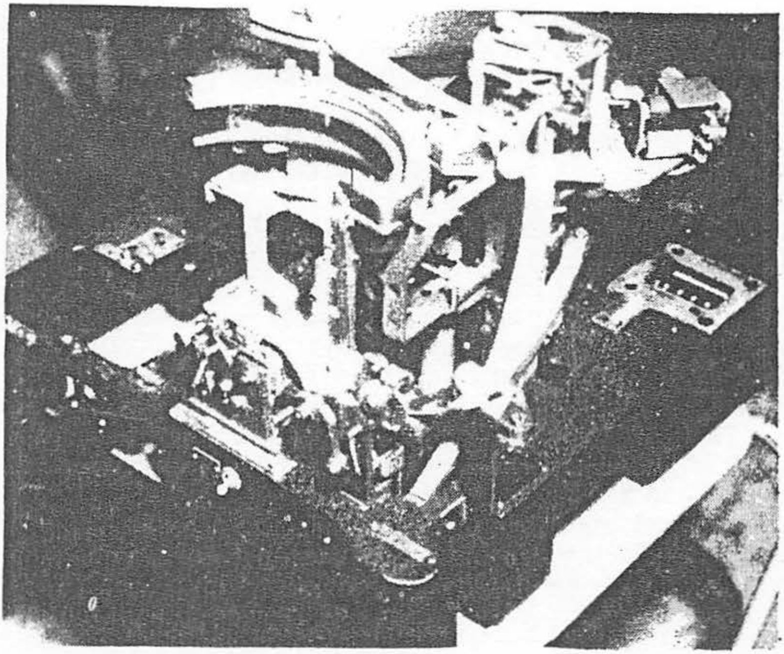


Fig. 3-1. The Model 14 typing reperforator.

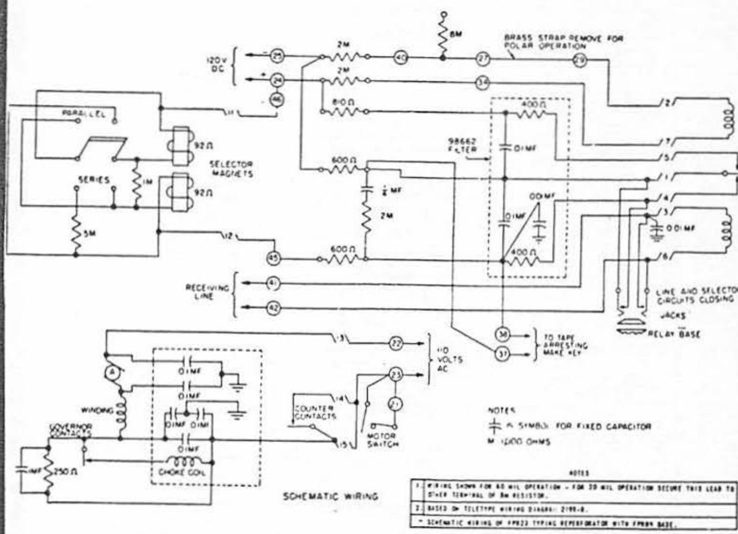


Fig. 3-2. Model 14 typing reperforator schematic.

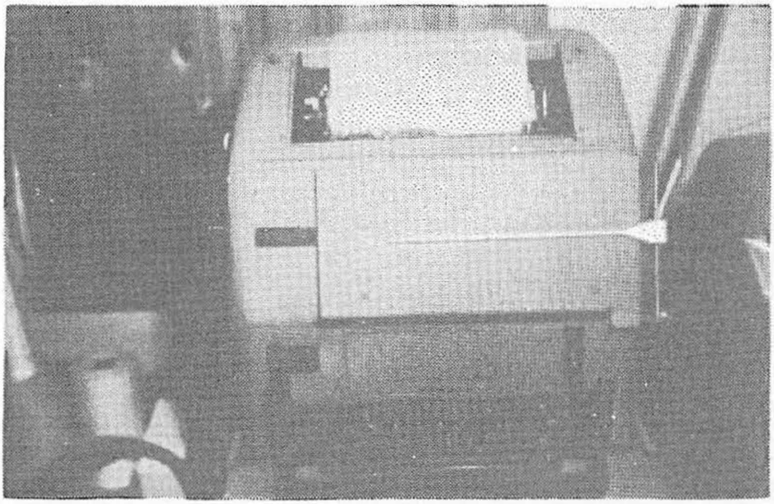


Fig. 3-3. Model 15 printer.

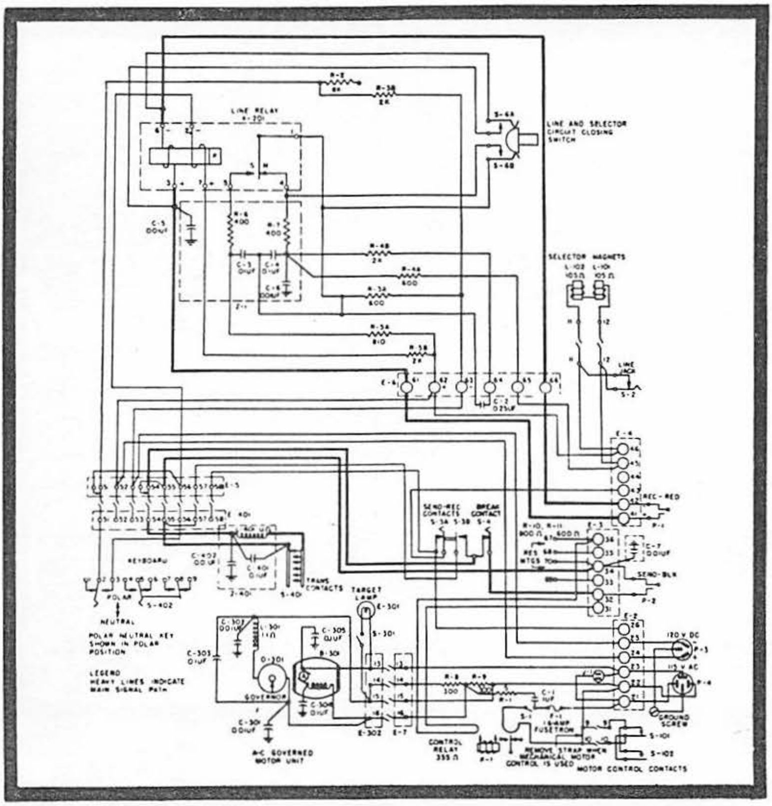


Fig. 3-4. Schematic diagram of the Model 15 printer and keyboard, base, and relay.

possible to punch tape while receiving incoming signals; and also, when the positioning switch in the front of the machine is set in keyboard-tape position, you can punch tape and print on the machine at the same time from your local loop. When this switch is in the tape position, only tape will be punched. In this position, unless your machine is equipped with automatic carriage return and line feed, watch your end of line indicator, which is a light that comes on at the end of each line. When the light comes on, return the carriage and hit your line feed key.

The Model 19 also includes tape transmitter located on the left side of the table. This is the piece of equipment which permits your transmitter to be keyed from tape at a speed of 60 wpm. A separate motor operates this unit and for operation, must be turned on by a toggle switch. The switch is located on the front of the tape transmitter. A master line switch on the

left side under the front of the table will cut off all AC line voltage.

The Model 19 is really about the last word in equipment for the hamshack. The only drawback is its high cost. Sometimes you can locate one of these if you shop around. There are many of them in hamshacks now so they must be available.

This model is illustrated in Figs. 1-1 and 2-1. Figs. 3-6 and 3-7 show the tape transmitter with cover removed and schematic. Schematics of additional equipment are shown in Figs. 3-8 and 3-9.

MODEL 26

This is one of the very popular printers (Fig. 1-2) for amateur use. They were originally built for the telephone company by the Teletype Corporation, but they didn't stand up as well as the Model 15 under 24-hour a day use, and were declared obsolete while still quite new. Fortunately, by this time the amateurs were well enough organized so that a great many of these machines were sold to RTTY'ers rather than the local junk men. The good supply of these machines has driven the price down to a very reasonable range.

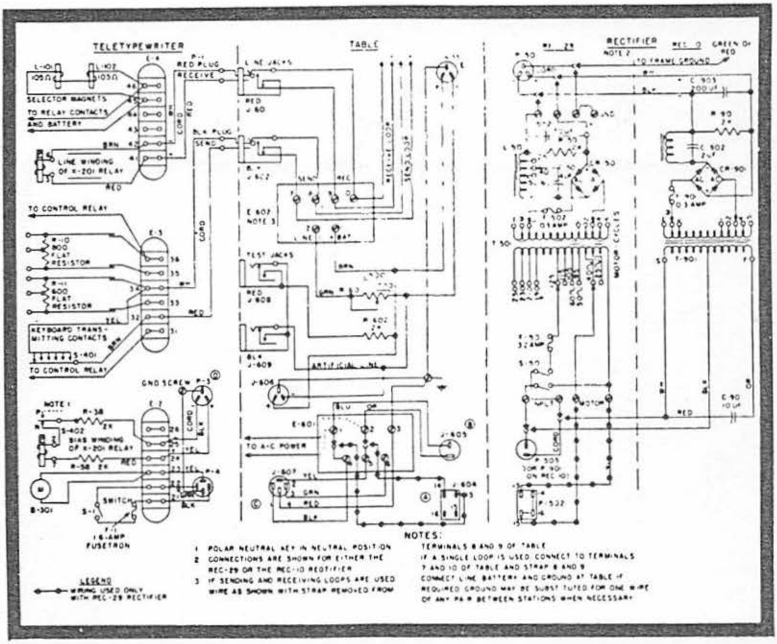


Fig. 3-5. Schematic diagram of the Model 15 table and optional rectifiers.

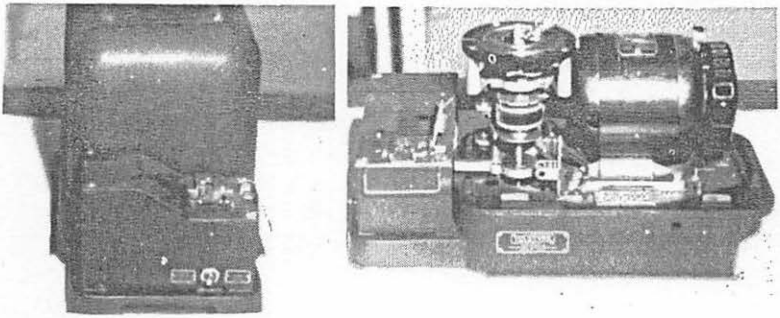


Fig. 3-6. Model 19 tape transmitter with cover removed.

The Model 26 is a single magnet printer, as are all but the old Model 12 and the very rare Model 21 receiving printer. It uses a small type wheel which is rotated by the distributor to print the letters on the page. The page stands still and the type wheel travels back and forth as in the Model 15 and the newer IBM typewriters. This is a fine machine for amateur RTTY.

MODEL 401

This is a receiving tape printer (Fig. 3-14) used by Western Union and is complete with an AC motor driven distributor. These units, when you can find them, are fine for a

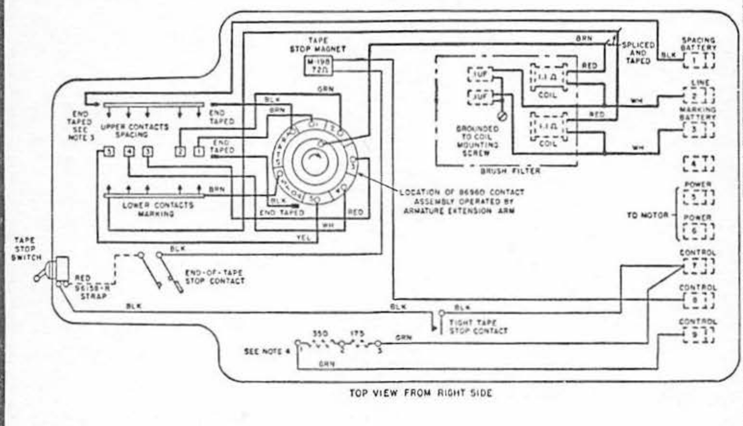


Fig. 3-7. Schematic of Model 19 tape transmitter.

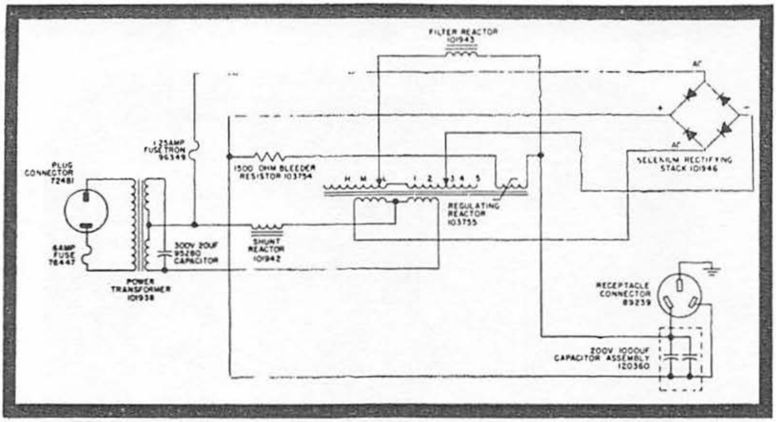


Fig. 3-8. Schematic diagram of Teletype Corporation Model REC-13 rectifier, used in some variations of the Model 19 and in other equipment.

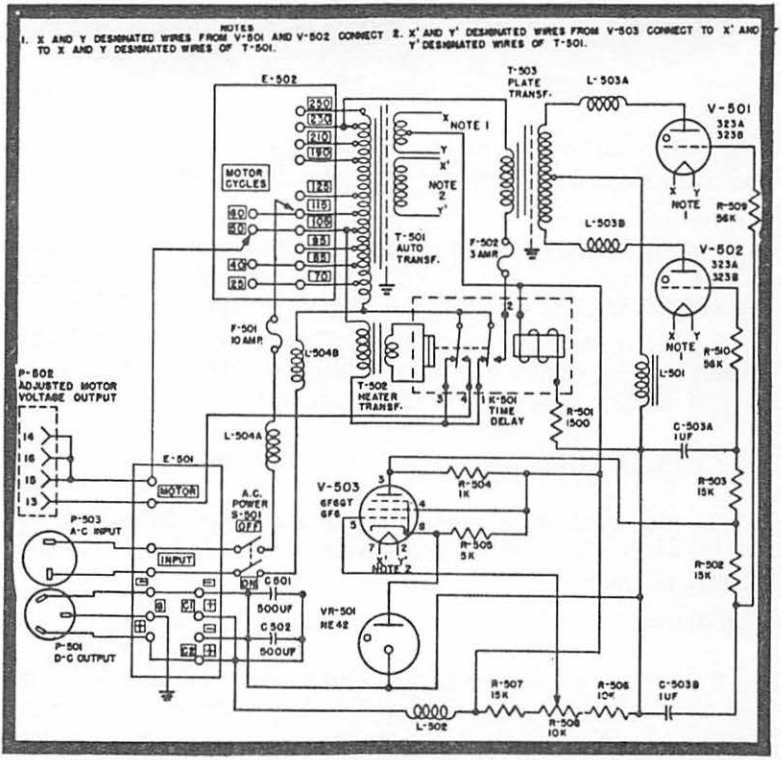


Fig. 3-9. Schematic diagram of Teletype Corporation Model REC-30 rectifier, used in some variations of Model 19 and other equipment.

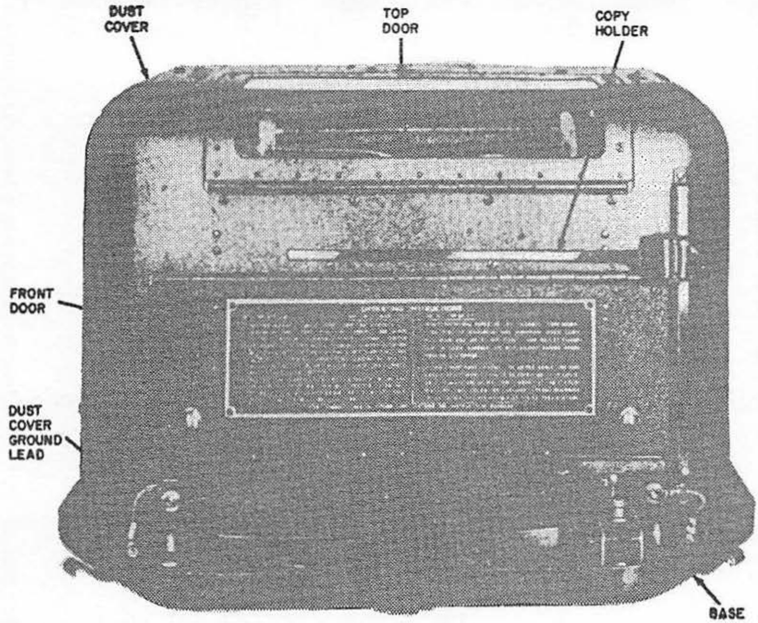


Fig. 3-10. Teletypewriter TT-4-G lightweight, portable Kleinschmidt page printer used by the Armed Forces.

remote repeater or even a regular station printer if you use a lot of tape transmitting. You can let this printer print the off-the-air signal while you cut tape with your regular printer.

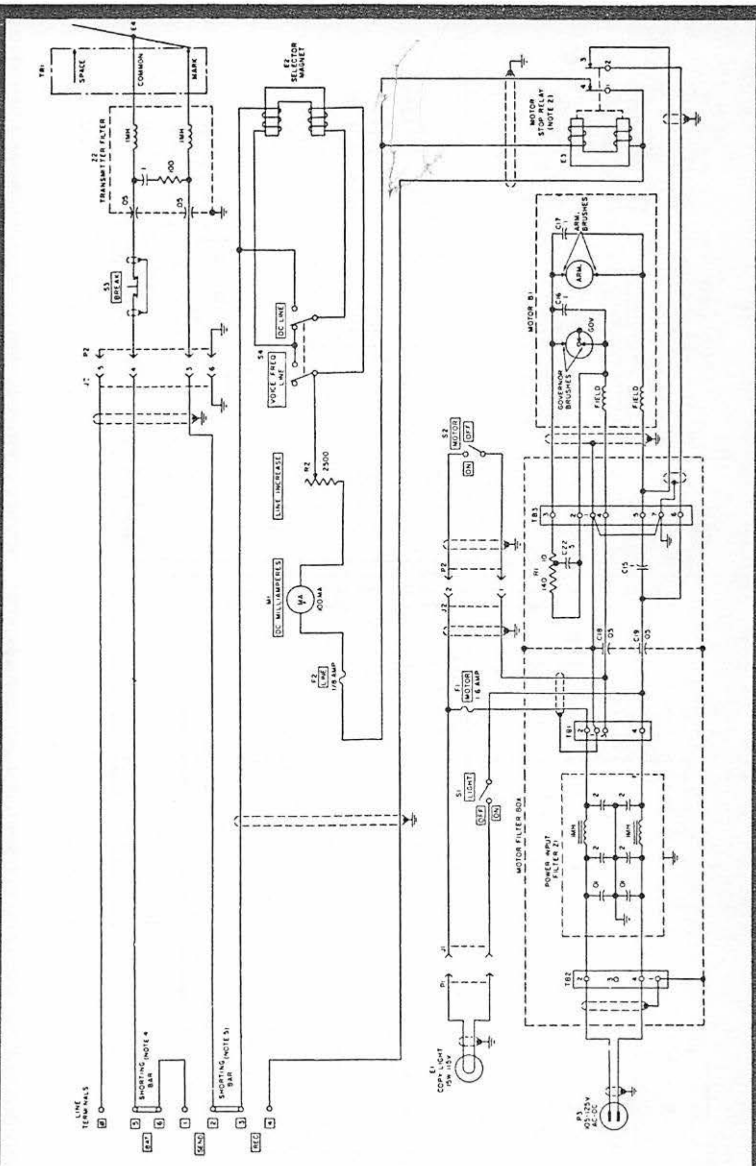
The 401 prints on the standard Western Union five thirty-second inch tape from a small type wheel, similar to the Model 26. The whole unit is less than one cubic foot in size.

MISCELLANEOUS ADDITIONAL EQUIPMENT

Various equipment and schematics are presented in Figs. 3-15 through 3-24 to provide a general view of the variety of models in use.

MODEL 31A

This is one of the newer machines (Figs. 1-4 and 3-13) and was designed for portable use. It has a keyboard almost identical to the Model 28 and an AC-DC motor for complete flexibility. The magnets only require half as much current as larger printers, though 50 or 60 ma doesn't seem to cause any difficulty. There are three cables; line cord for the motor,



NOTES,

1 UNLESS OTHERWISE INDICATED RESISTORS ARE IN OHMS, CAPACITORS ARE IN UF

2 MOTOR STOP RELAY CONTACTS LATCH MECHANICALLY IN CLOSED POSITION AND OPEN BY OPERATION OF MOTOR-STOP MECHANISM. ONCE OPEN, CONTACTS REMAIN SO, AS LONG AS RELAY IS ENERGIZED. CONTACTS CLOSE AND LATCH AGAIN UPON BREAK AND MAKE OF LINE CURRENT

3 END VIEW OF PLUGS AND JACKS

4 SHORTING BAR CONNECTED ACROSS LINE TERMINALS 5 AND 6 WHEN POWER FOR SIGNAL CIRCUIT IS NOT SUPPLIED LOCALLY. DISCONNECT SHORTING BAR WHEN USING LOCAL POWER SOURCE

5 SHORTING BAR CONNECTED ACROSS LINE TERMINALS 2 AND 3 FOR HALF-DUPLEX OPERATION. DISCONNECT SHORTING BAR WHEN INSTALLING FOR FULL-DUPLEX OPERATION

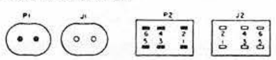


Fig. 3-11. Schematic diagram of the TT-4-G.

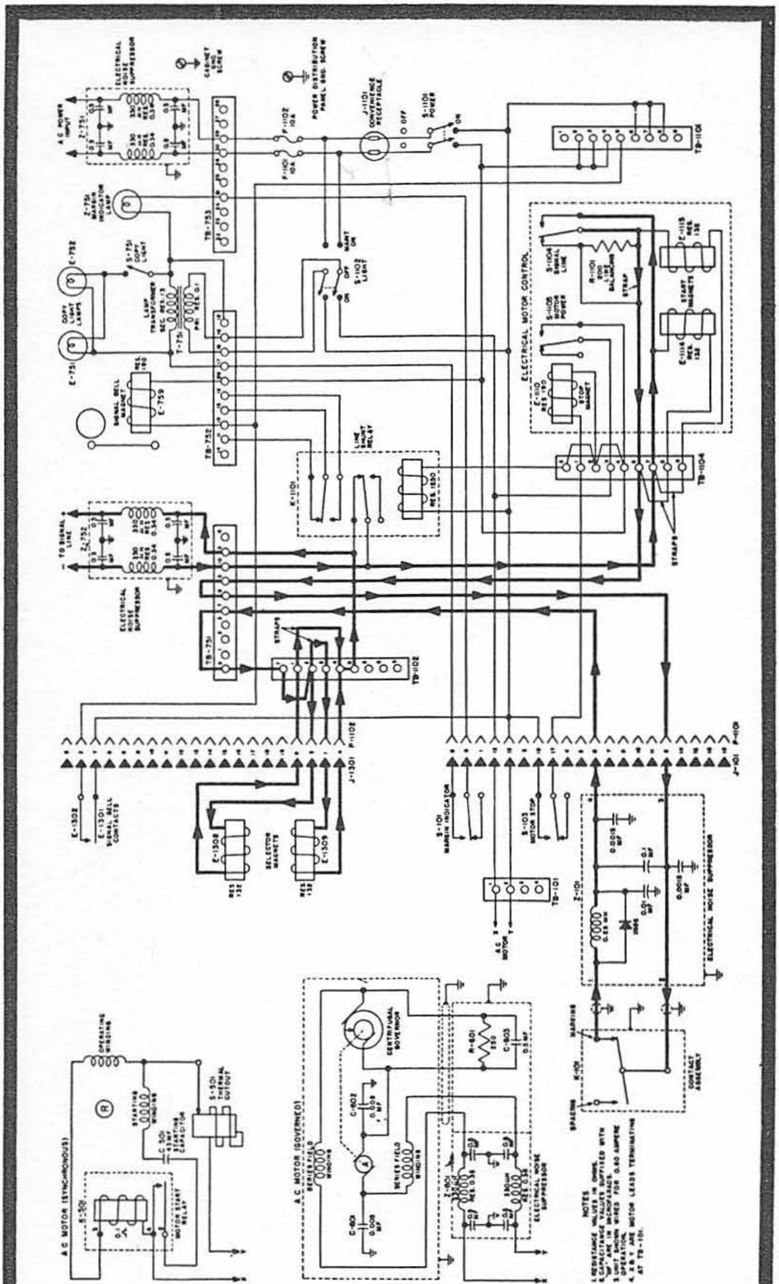


Fig. 3-12. Schematic diagram of Model 28. For illustration see Fig. 1-2.

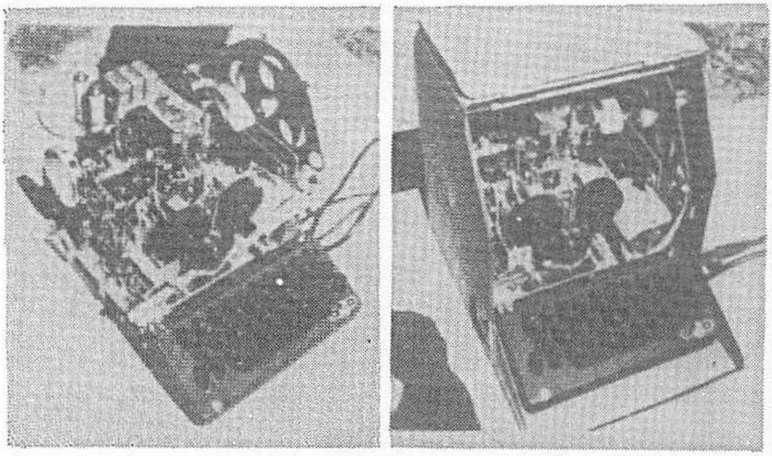


Fig. 3-13. Model 31A with cover removed. (See also Fig. 1-4.)

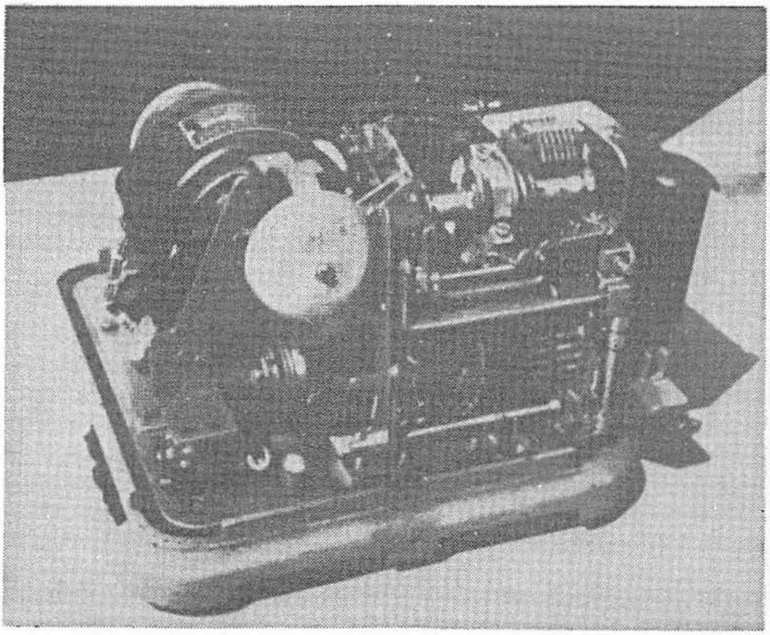


Fig. 3-14. Model 401 receiving tape printer.

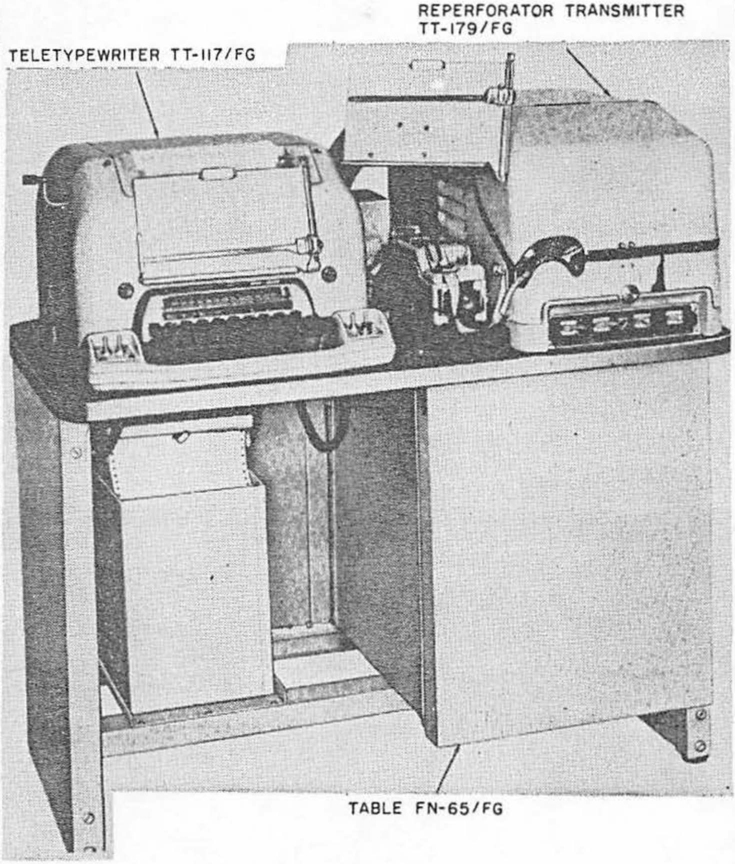


Fig. 3-15. Kleinschmidt Teletypewriter set AN-FGC-25 made for the Armed Forces.

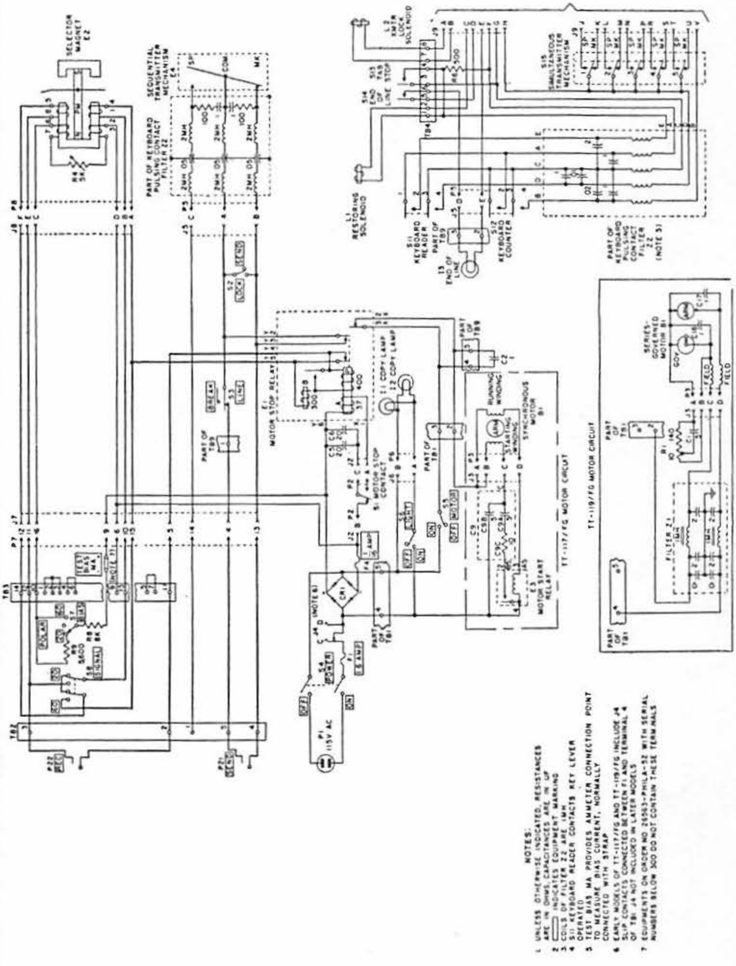


Fig. 3-16. AN-FGC-25 diagram.

(Cont'd. on page 60)

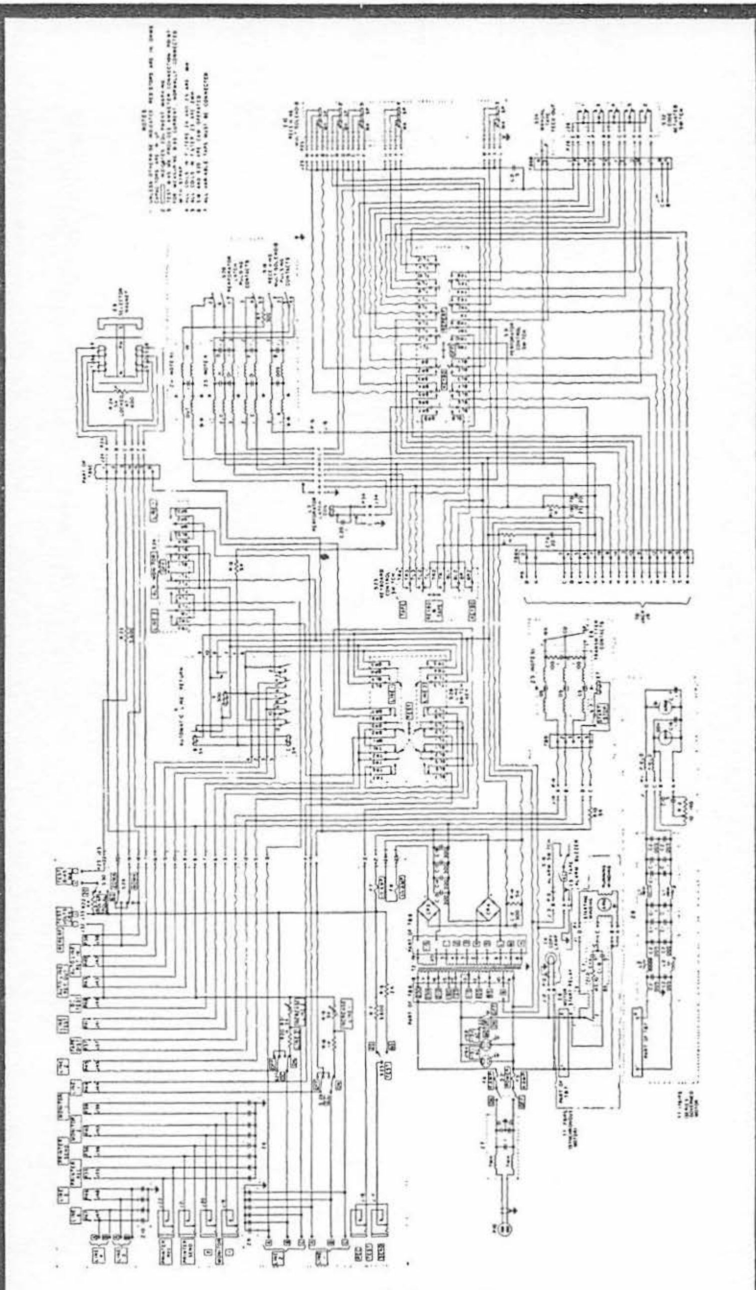


Fig. 3-16. AN-FGC-25 diagram.

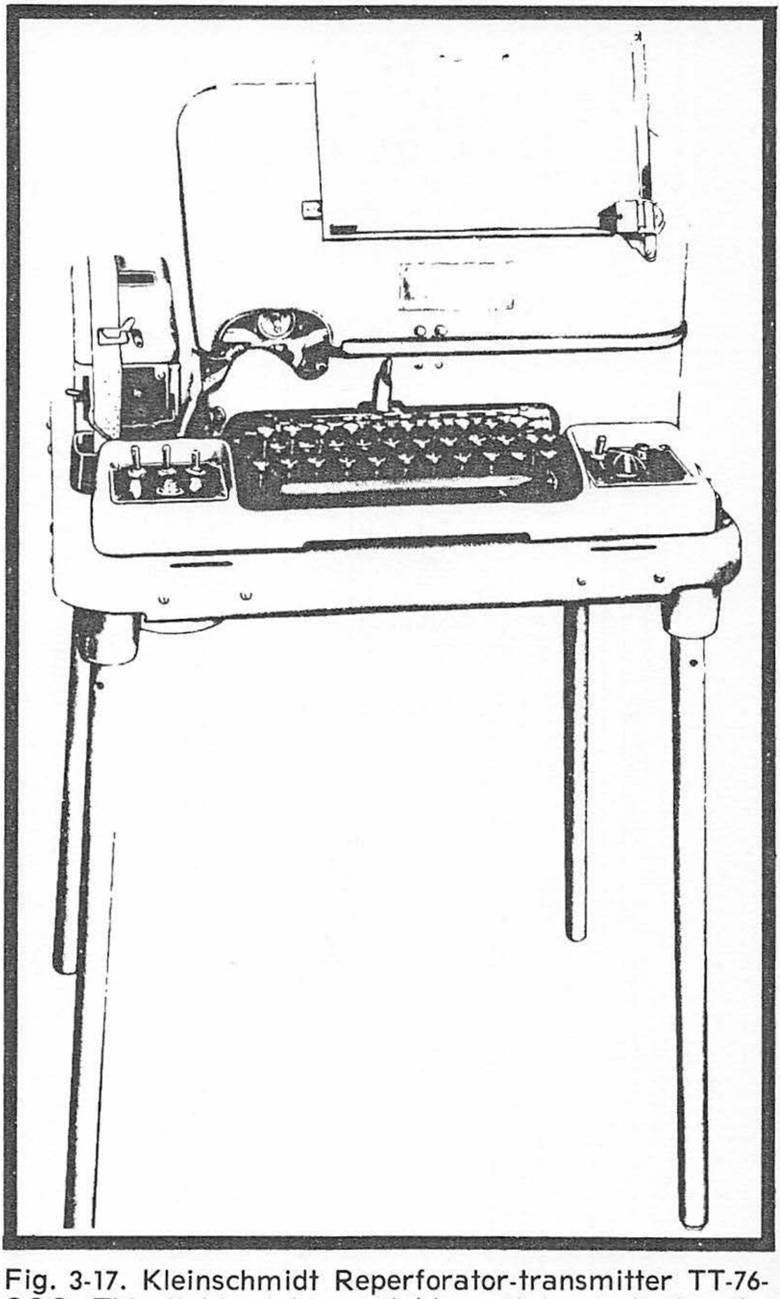


Fig. 3-17. Kleinschmidt Reperforator-transmitter TT-76-GGC. This lightweight, portable unit is made for the Armed Forces.

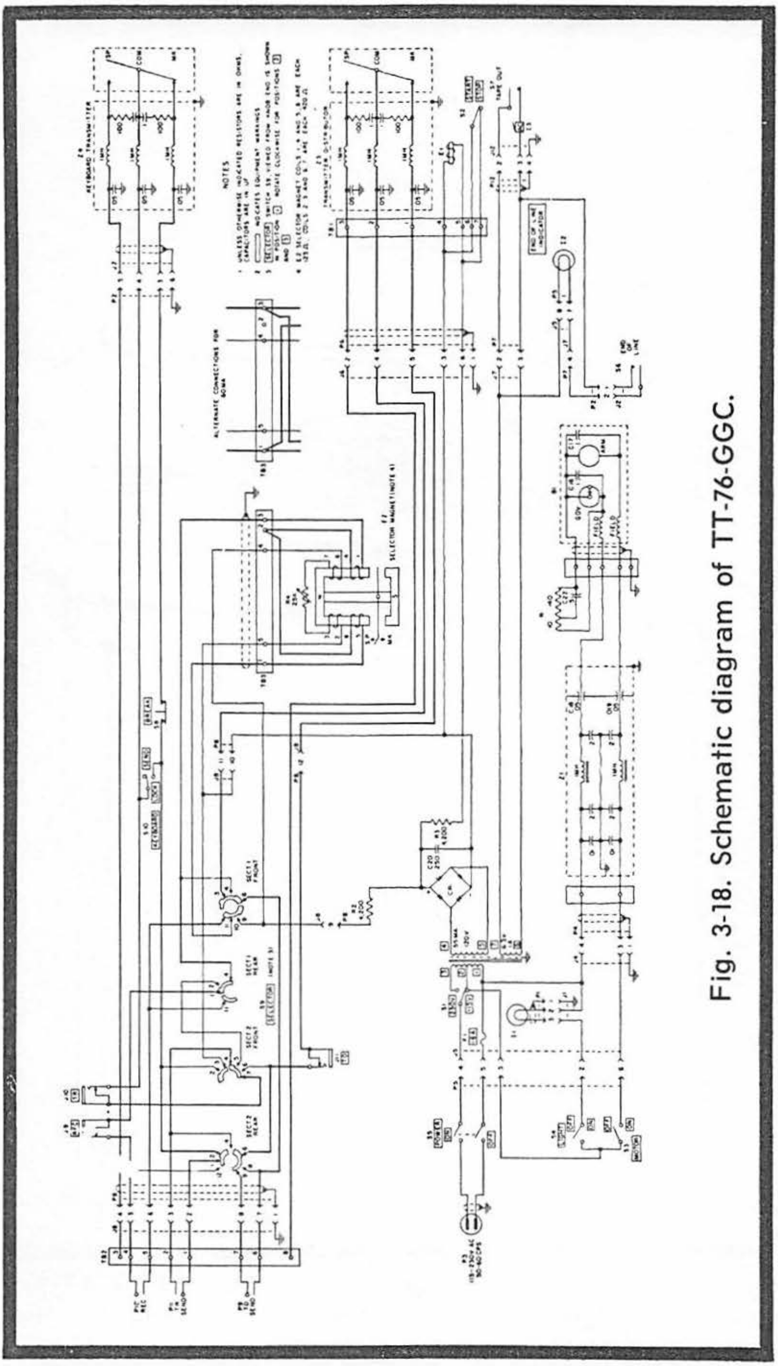


Fig. 3-18. Schematic diagram of TT-76-GGC.

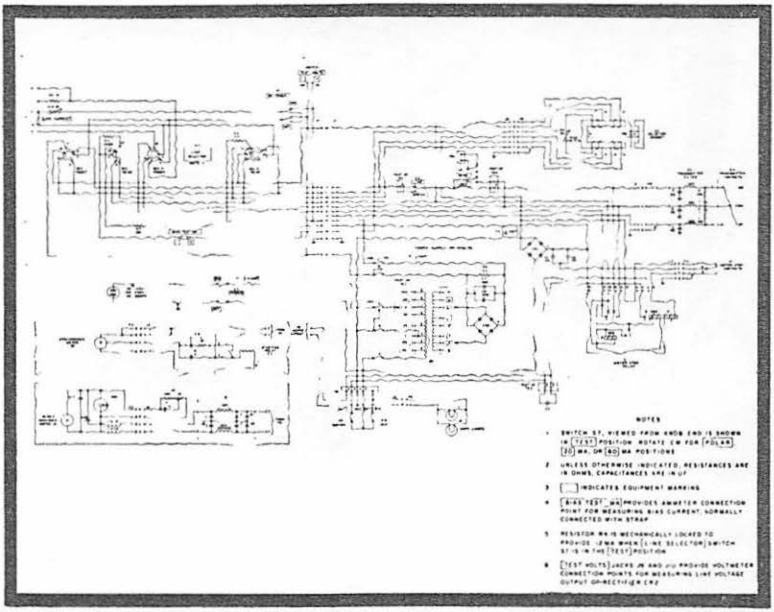


Fig. 3-19. Teletypewriter set AN-FGC-20 schematic diagram. For illustration see Fig. 1-5.

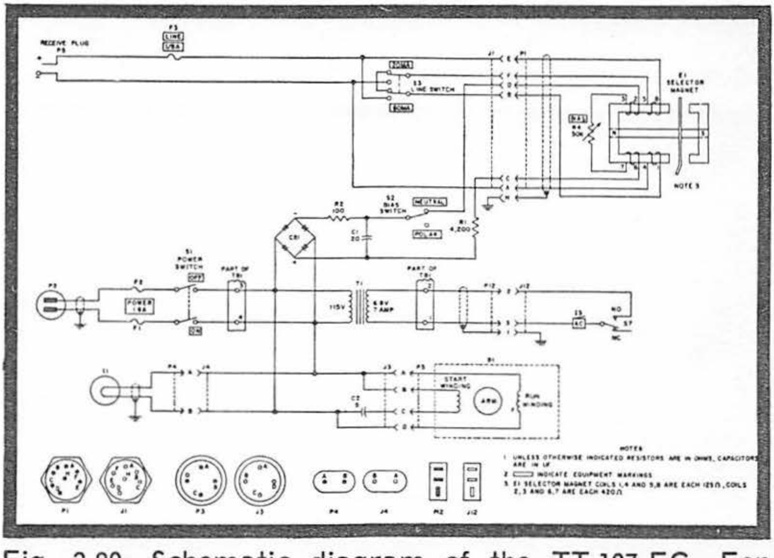


Fig. 3-20. Schematic diagram of the TT-107-FG. For illustration see Fig. 9-3.

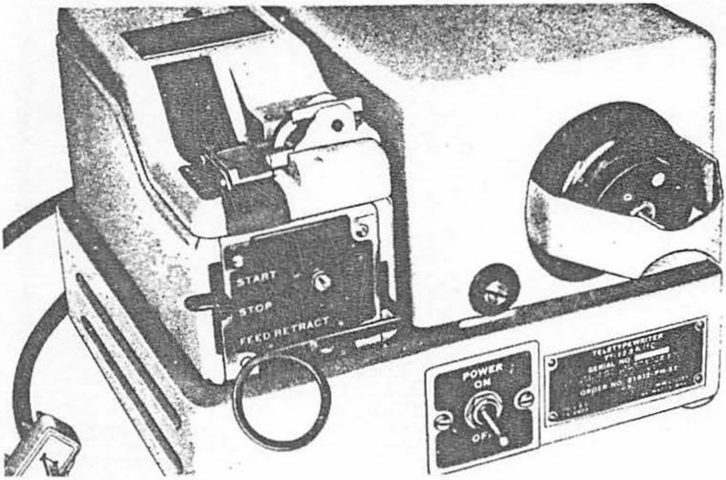


Fig. 3-21. Model TT-123A-FG.

keyboard, and printer. This machine was made by the Teletype Corporation and used by Western Union. It is very light, weighing about 19 pounds complete with aluminum case. The governed motor, as the machine comes, is a bit noisy and has to be filtered for good radio operation. The 31A uses five thirty-second inch tape and has a built in end of line indicator which is quite important when you are in contact with page type printers. The light lights when you are near the end of the line and you can send the carriage return and line feed signals.

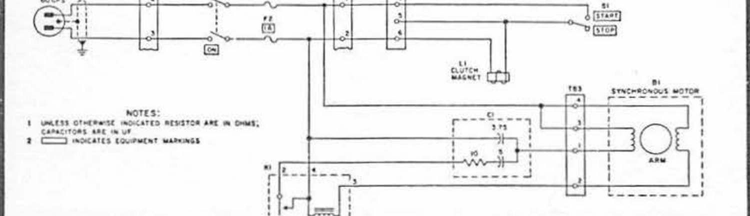


Fig. 3-22. Schematic diagram TT-123A-FG.

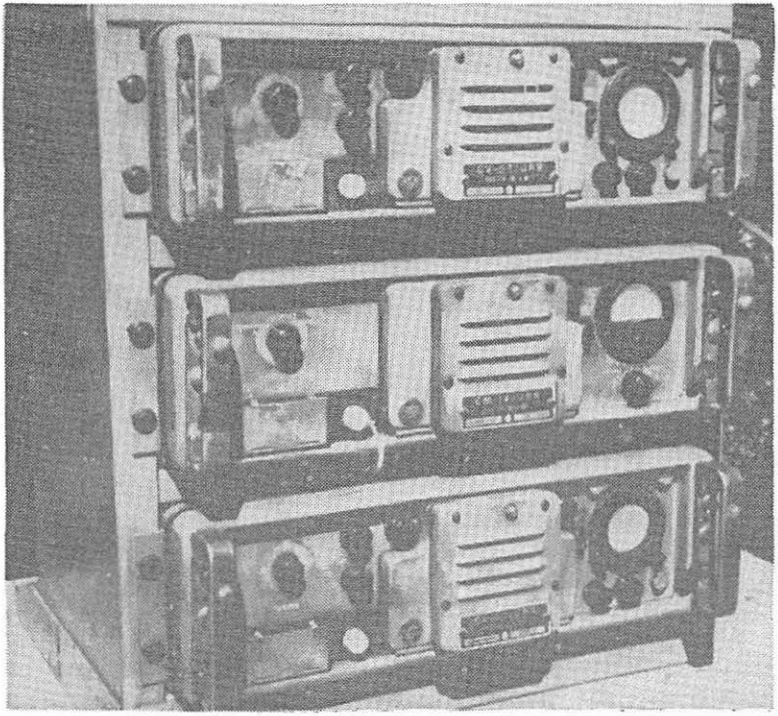


Fig. 3-23. URC-6 units.

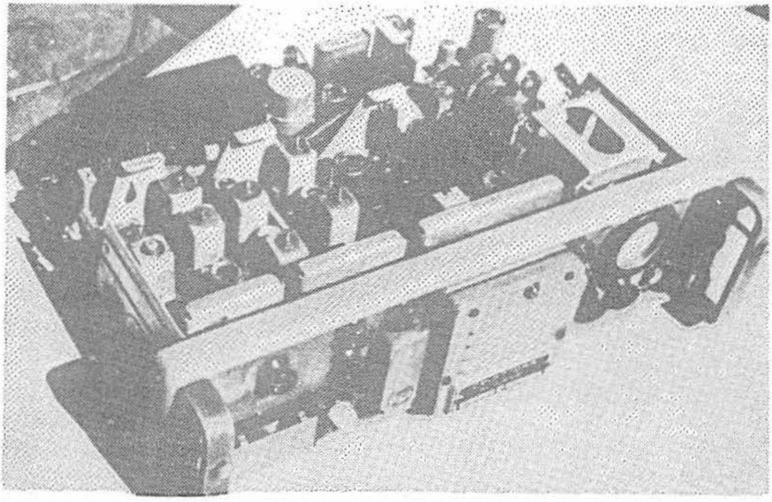


Fig. 3-24. Schematic diagram URC-6.

CHAPTER 4

Terminal Units

Once you have a machine you need a converter to complete your receiving setup. Let's look at the converter situation and see just what is required. There are two types of converters. In either type the main function in the overall circuit is to change incoming signals into DC pulses that operate the selector magnets on the machine.

A SIMPLE TUBE-TYPE CONVERTER

We will deal later with the technical aspects of converters. For the present, here is a simple converter (Fig. 4-1) that does an excellent job. The circuit in Fig. 4-2 uses only the necessary parts to perform the function and all frills omitted.

This unit employs a progressive type of construction. That is, the converter is wired in one complete section. The power supply is constructed as another unit. A monitor scope, which can be added at any time, is built as another section, and all three are removable from the overall chassis for servicing or adding additional circuits as required.

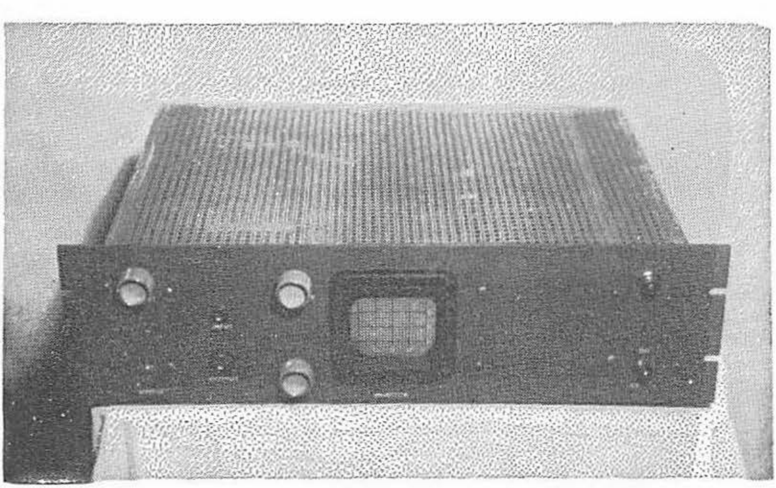


Fig. 4-1. A tube-type RTTY converter.

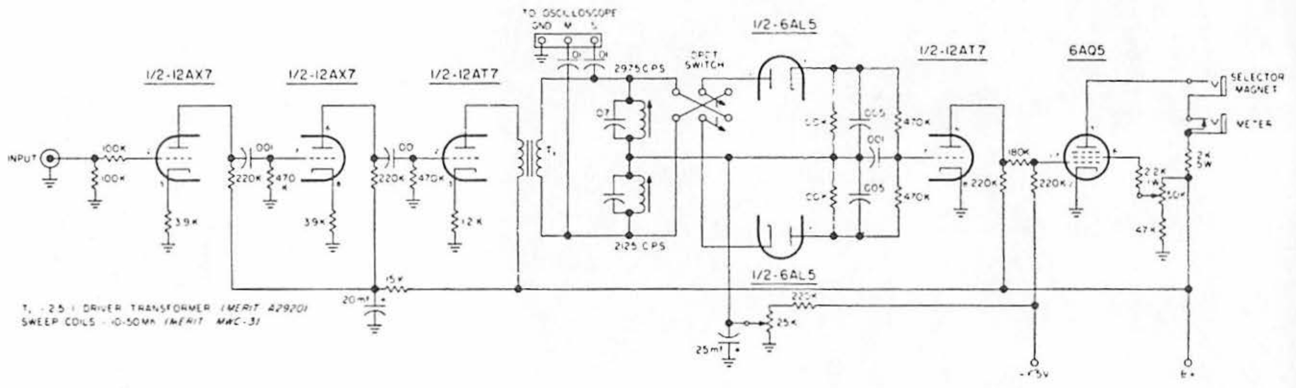


Fig. 4-2. Schematic diagram of tube-type converter.

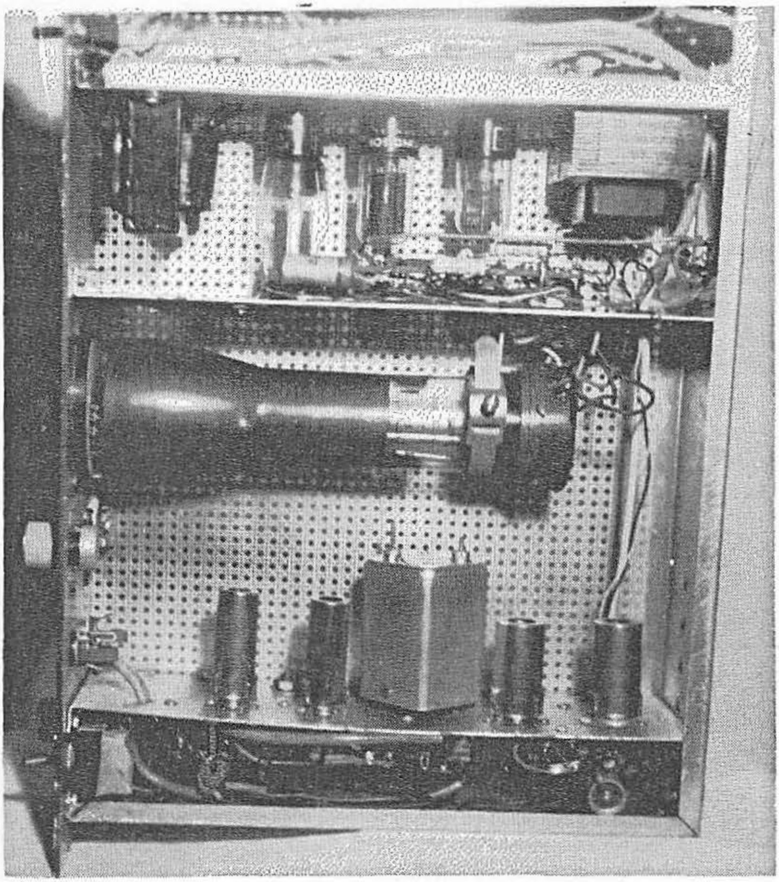


Fig. 4-3. Internal layout of tube-type converter.

Construction

For those interested in the construction details, a 4" x 13" x 17" aluminum chassis was used. Cut out the top, leaving a lip all around of ½ inch. The section cut out is then cut to make two sub-chassis; one for the converter and the other for the power supply. A strip of light aluminum will be required to make the third sub-chassis for the monitor scope.

Each of these sub-chassis is cut to allow a ½-inch bend at each end and still fit against the front and rear wall of the overall chassis. They are held in place with small aluminum self-tapping screws.

The whole assembly is mounted on a 5¼" by 19" standard relay rack panel. The top and bottom are covered with per-

forated aluminum which is attached to the edge all around with aluminum self-tapping screws.

As to the converter itself, there is nothing new about it. It has been used by RTTY'ers for some years in this or some modified form. It was selected because it is easy to assemble (Fig. 4-3) and uses parts that are readily available. As can be seen from the schematic, the wiring is easy; and, since there is nothing critical about it, it should work the first time it is hooked up.

Be certain that the meter jack is of the closed circuit type. The output jack is open when the plug is removed. Both of these jacks are insulated from the chassis since there are about 200 volts present in the circuit at this point.

Adjustments

A 100 milliampere meter should be inserted in the meter jack for making adjustments. It may be left in the circuit after adjustments are made or may be removed, since the closed circuit jack will close and current will pass.

In the final adjustment of the converter turn the control bias down to zero and the current adjustment down to minimum screen voltage. With your machine connected at the output terminals of the converter, apply line voltage; and turn on the power.

After warmup, without any audio input being applied to the converter, there will be no output current indicated. At this point feed a 2975 Hz audio signal to the input and adjust the coil with the .07 mfd condenser across it for maximum indication on the scope. The scope is connected to the scope terminals on the converter.

Reset your signal generator to 2125 Hz and adjust the other coil slug for maximum scope presentation. This presentation will be at right angles to the first.

With no audio input, start your machine and adjust the bias control until the loop current reaches a steady value. It should read about 18 or 20 ma with the current adjust pot set to minimum.

Adjust the current control for the printer current desired. You may find that you will need to back off on the bias control slightly and reset it until the magnets pull in.

The meter should read about 30 ma at this point. When a signal is fed to the input of the converter, the meter current will show a sharp swing up to about 50 ma, and will continue this swing as the signal is received.

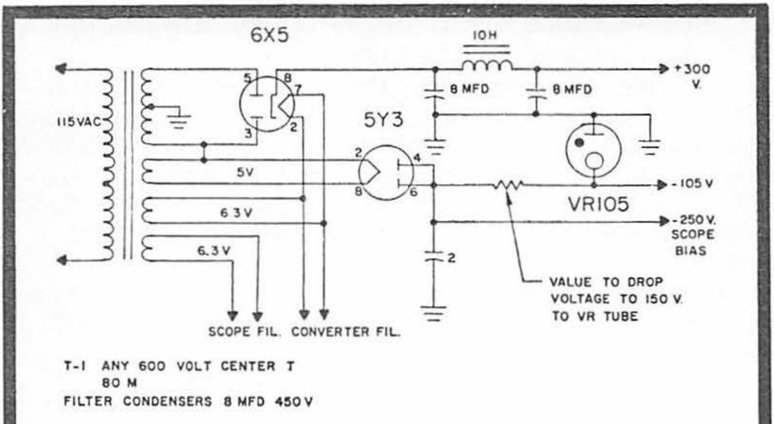


Fig. 4-4. Power supply for tube-type converter.

A few tries will give you the proper adjustment know-how. You will soon find the machine rattling away with perfect copy received.

The power supply shown in Fig. 4-4 is designed to provide power for both the converter and the monitor scope. Voltages indicated should be closely followed, since, if they are much different, the DC amplifier in the converter will not operate correctly. As a result, the converter will not function properly.

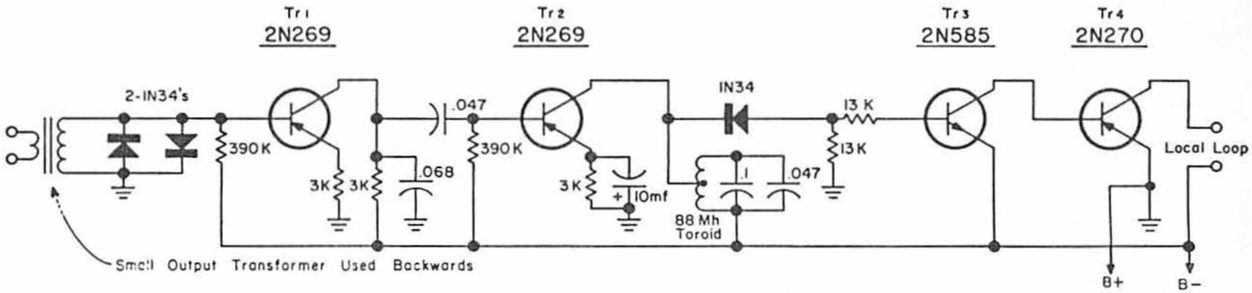
A BASIC TRANSISTORIZED RTTY CONVERTER

The circuit of Fig. 4-5 is about as simple as possible to construct and still obtain good readable copy on the ham bands.

Audio from the receiver is fed through T1 and is limited to a set value by the 13K resistor and the two 1N34's connected across the secondary. The first 2N269 is self-limiting by virtue of the unbypassed resistor in the emitter circuit. Also, there is a fairly large value bypass condenser connected between the collector and ground of this same transistor: the purpose of this is to reduce high frequency noise and signal leak-through.

The second 2N269 is utilized as an amplifier and provides more than enough gain to drive the 1N34 rectifier. The collector of this transistor is connected to the center tap of the toroid rather than the outside. This is to reduce the loading effect of the collector impedance on the selectivity of the tuned circuit.

Transistor TR3 acts as a DC amplifier and builds up the output of the rectifier to a value high enough to key the output



13K Resistor across secondary of transformer not shown in schematic.

Fig. 4-5. Schematic diagram of a basic transistorized RTTY converter.

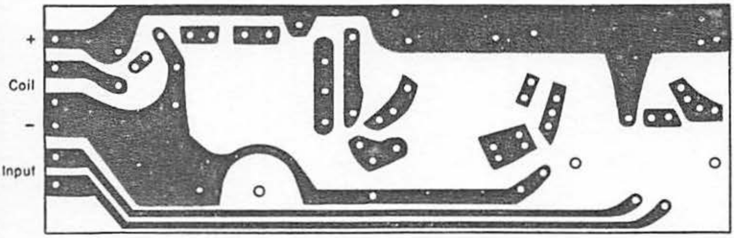


Fig. 4-6. Circuit board for basic transistorized converter.

transistor. The two 13K resistors in the base circuit of this transistor are the only really critical parts of the unit. Do not change the value of these as they set the current of the output loop. Also, it is easily possible to destroy transistor TR3. If you must experiment with these values, connect a milliammeter in the emitter circuit of TR3 and watch the value of this current closely. Also be sure that the 1N34 rectifier is connected to the junction of the two resistors, not to the upper end, as this also controls the amount of current drawn by this transistor.

With a 12- to 15-volt supply connected, the local loop current will be between 60 and 65 ma and can be used to key the machine magnet directly. The tuned circuit is set to the space frequency; and, when a signal is received, the current in the local loop drops to almost zero.

This unit is built on a home-made printed circuit board. The layout is shown in Fig. 4-6. The full size dimensions are 3" x 9". For connection to the external circuits, use bolts and nuts, with the heads of the bolts soldered to the printed circuit.

AN ALL SILICON TRANSISTOR RTTY CONVERTER

A TU should be as simple as possible to build and adjust, and should cost as little as practicable. It should be usable on FSK or AFSK. It should have provisions for filters such as bandpass and narrow shift. It should provide for some sort of tuning indicator (meter, oscilloscope, or tuning eye) and should incorporate a provision for auto-start and an AFSK oscillator for VHF.

The silicon transistor is a good type of device to handle the current required, which is about 200 ma. The TU (Fig. 4-7) is built on a printed circuit board.

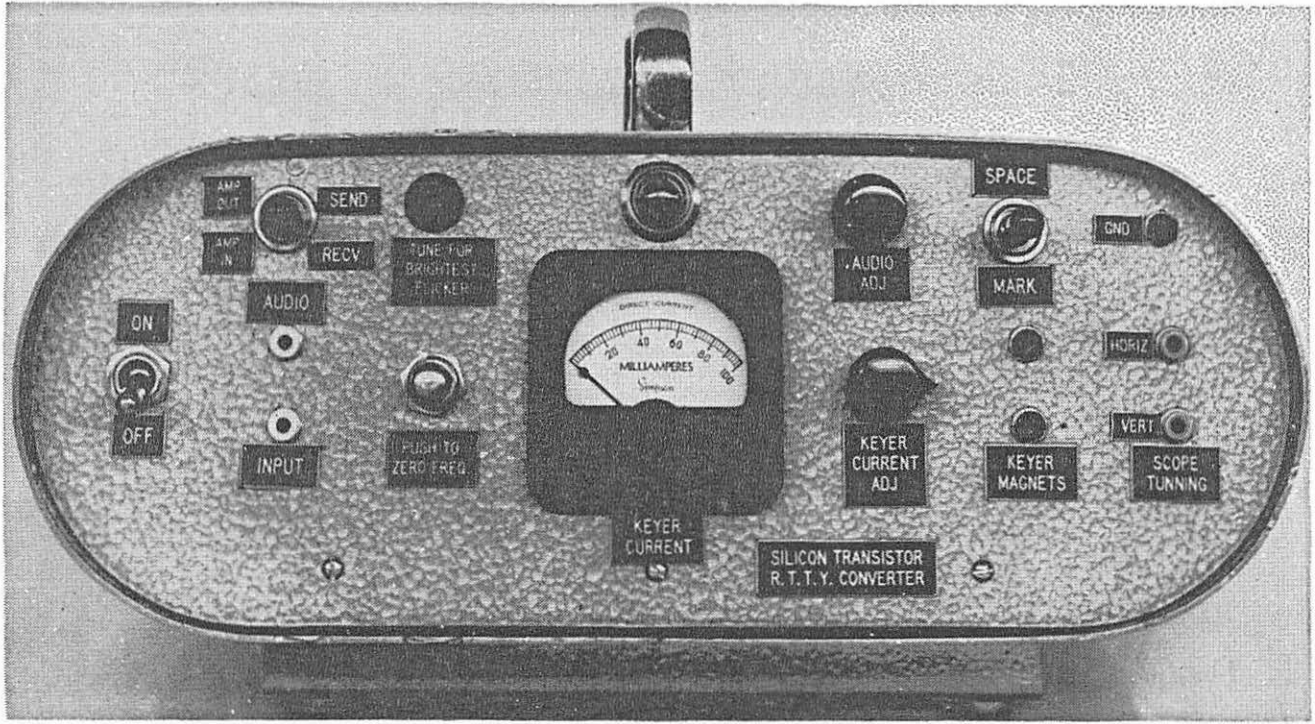
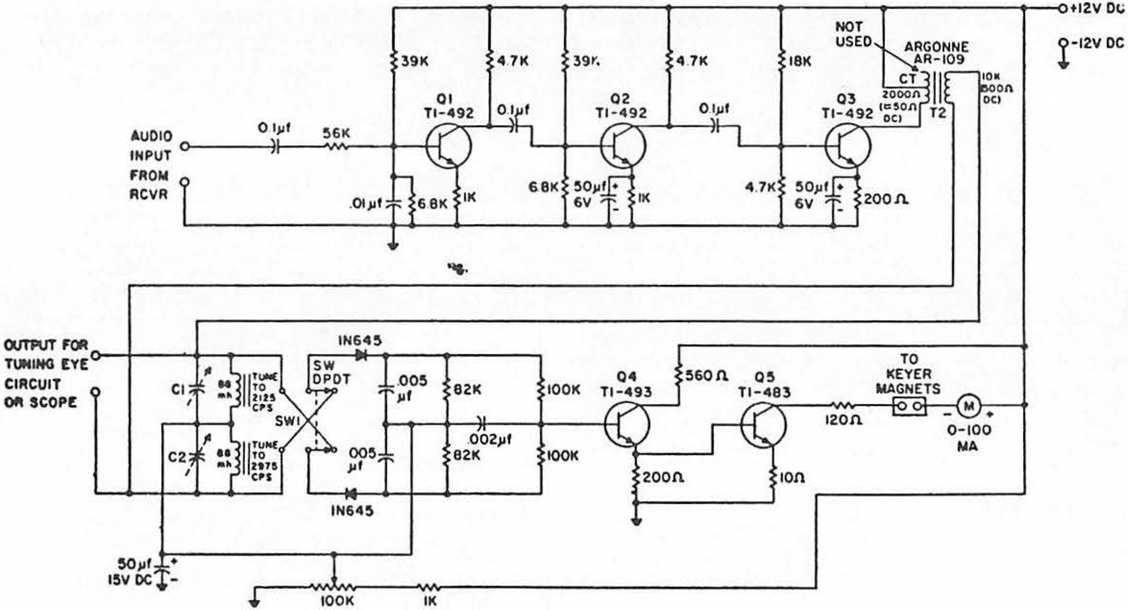


Fig. 4-7. An all-silicon-transistor RTTY converter.

Fig. 4-8. Schematic of all-silicon-transistor converter.



Schematic of converter. C1 will use approximately .068 mfd to tune to 88 mh choke to 2125 cycles. C2 should tune to 2975 cycles with about .033 mfd. T2 can be a Stancor TA-35, a Thordarson TR-7, or an Argonne AR-109.

Adjustment is easy and requires only an audio oscillator and VTVM to adjust the two 88-mh toroids.

An LC filter or bandpass amplifier can be added to the front of the converter to improve reception when there is heavy QRM.

The provisions for tuning have been made for an oscilloscope, a tuning eye, or frequency meter. (See Fig. 4-8.)

The circuit construction is simple (Fig. 4-9). The circuit board (Fig. 4-10) has numbers for each lead of the components

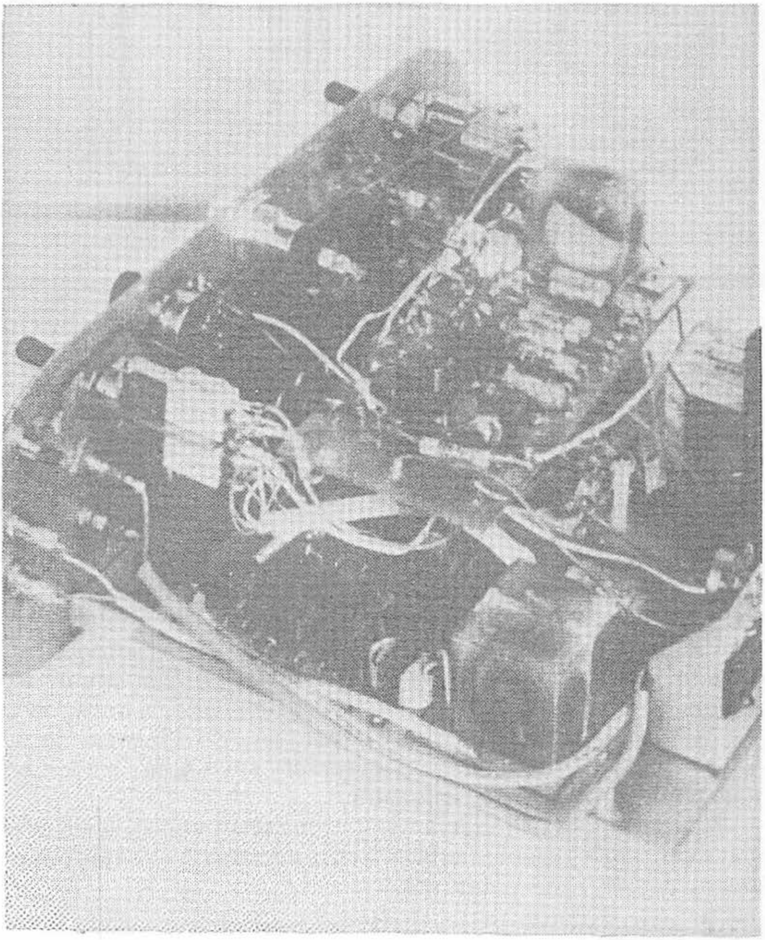


Fig. 4-9. Internal arrangement view of all-silicon-transistor converter.

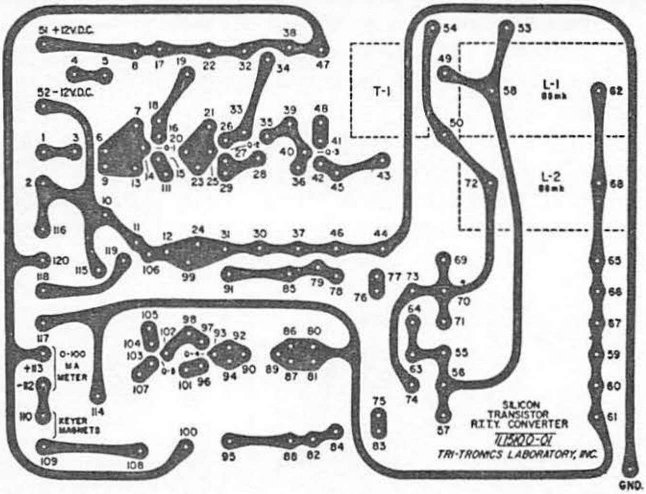


Fig. 4-10. PC board for all-silicon-transistor converter.

to be soldered. There are six holes to provide for as many as three capacitors, to tune each of the 88-mh toroids for the standard 850 cycle shift, the space filter is tuned to 2975 Hz. This takes a total capacity of about 0.033 mfd. About 0.068 mfd capacity is used to obtain 2125 Hz with the other toroid. NOTE: Use only mylar, mica, or paper capacitors. Do not use disc ceramic capacitors. When tuning the two toroids, they should be mounted on the board and in the circuit. An audio signal is applied to the input at the frequency each of the toroids is to be tuned to. Watch for a peak on a VTVM across the toroid being tuned. The input circuit shown is an RC type input into the base of the first transistor. A transformer may be coupled between the output of the receiver to match the input impedance of the converter. Although the converter works very well with four ohm input from the receiver, it will work much better with a matching transformer of four ohms to twenty thousand ohms into the converter.

The output of the converter is placed in series with the local loop. The output transistor acts as a switch or relay, and does away with the polar relay.

Parts Notes

All parts are available from your scrap box or your local dealer. The transistors and diodes are from Texas In-

struments, Inc. These silicon transistors are industrial type and do a good job for a reasonable price.

A Hybrid Version

If the transistors prove hard to find, or too expensive, a hybrid unit may be built as follows.

Using the same board, replace transistors Q1, Q2, Q3 and Q4 with 2N697 silicon transistors which cost much less. Replace Q5 with an RCA 2N3440.

When the unit is wired do not connect to terminal numbers 110, 112, 113, and 117; disregard them. For the power to the unit, use 12 volts DC as called for, but for the loop DC to the keying transistor (the 2N3440) use the power supply, shown in Fig. 4-11.

A Triad R-30 transformer may be used for T-3. It's rated at only 50 ma, but it will give 60 ma easily. It also has a 6.3 volt AC center tapped winding at 1.5 amperes, if you need it. A Stancor P-8421 may be used, or a Triad N-51X (an isolation transformer, rated at 130 volts DC at 300 ma). Any one of these three are acceptable.

Connect all parts as shown in the schematic. Insulate all three keying jacks and the B+ from the chassis. Connect the B+ to terminal number 109 on the board.

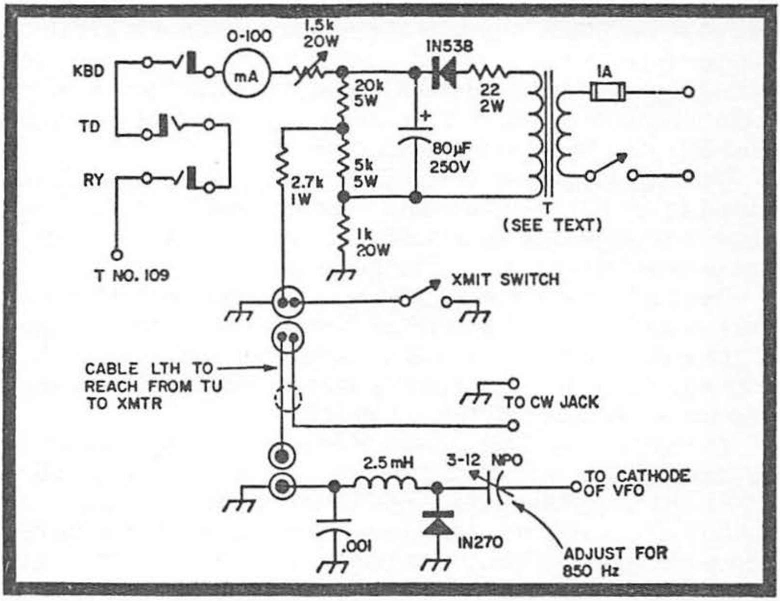


Fig. 4-11. Hybrid version of the "all-silicon-transistor" converter.

When the terminal unit is complete and wired, plug in the transmitter distributor (TD). If you do not own a TD, use a shorted phone plug in this jack. Plug in the keyboard and the printing relay. Turn on the unit and adjust the 100K pot connected to terminal 116, 118, and 120. Watch the 0 to 100 ma meter and stop at 60 ma. With no signal into the terminal unit tune in an RTTY station and begin to copy. If you get no copy or it is garbled, reverse the switch.

When you get tired of just copying, build an FSK board. Use a 3-circuit mike jack and a SPST switch to FSK the transmitter and to control it in the CW position. The TD, the KBD, or the incoming signal will key the transmitter.

AN RTTY TERMINAL UNIT USING IC'S

The basic unit consists of four functional blocks: the isolation amplifier, ratio detector, the voltage comparator, and the high voltage switch. The isolation amplifier and the voltage comparator are both low cost integrated circuits, whose use keep the cost of this unit to a minimum.

This terminal unit (Fig. 4-12) utilizes a UA-710 integrated circuit. This is a high gain, high speed, voltage comparator. Its operation can be summed up by stating that a reference voltage is applied on one input and the signal in question is applied to the other. Any time the incoming signal is greater than the reference, the output is in one state, and when the signal drops below the reference point, the output will switch to the other state. There is a very narrow threshold which gives this unit its extreme sensitivity.

The detector portion of this TU is a ratio detector, such as is used in an FM receiver. This type of detector is very insensitive to amplitude modulation which allows the FM (FSK) to be detected when it is down in the noise.

The incoming detected signal is integrated across C5 which is divided by R7 and R8 to form the reference voltage for the comparator. Since this voltage is proportional to the detected signal, it automatically tracks and eliminates the need for a variable reference control.

T1 and T2 (Fig. 4-13) consist of the standard 88 mh toroid telephone chokes which are available surplus or as generally advertised. They have been modified into transformers by the addition of 50 to 100 turns for a secondary. The number of turns used is not at all critical, as long as the same number of turns are on each transformer.

A-1 is used as an isolation amplifier to drive T1 and T2 differentially, and is a Darlington differential amplifier. The

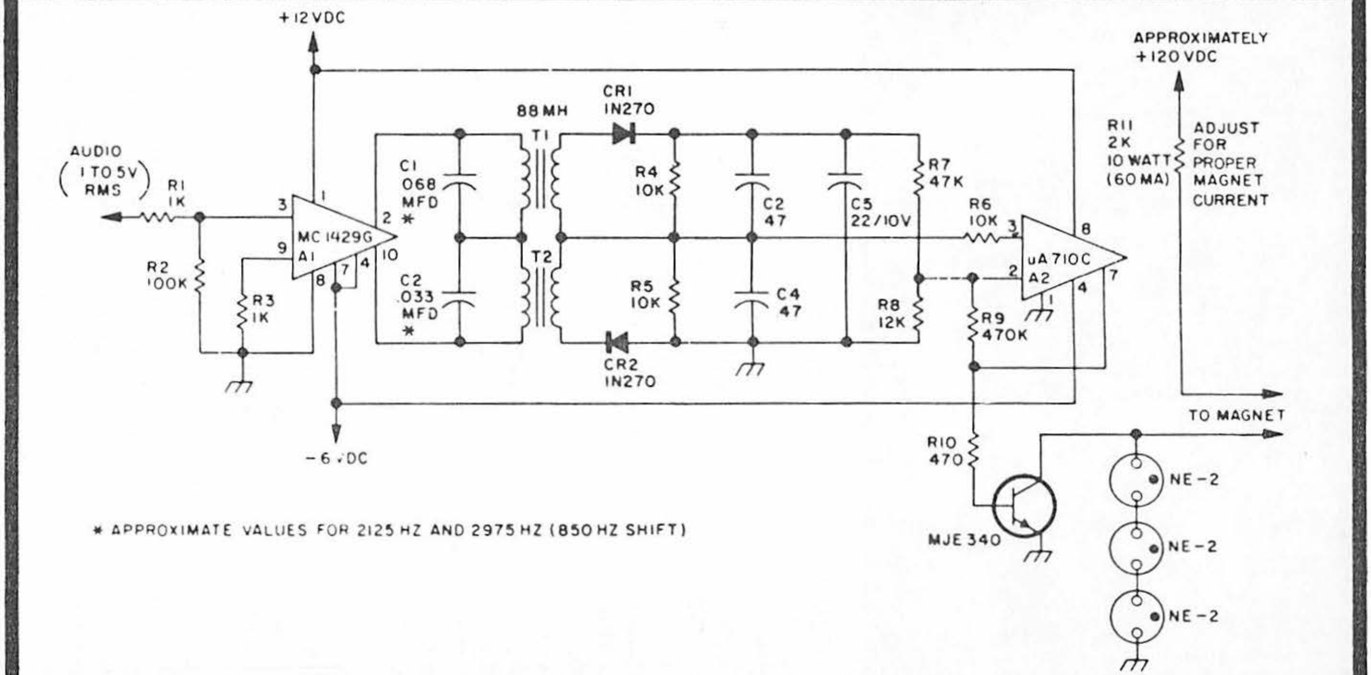


Fig. 4-12. Schematic diagram of a RTTY audio terminal unit.

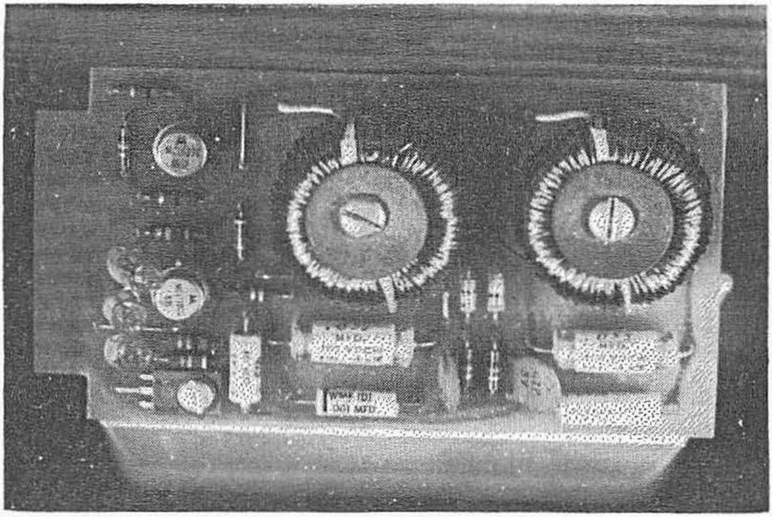


Fig. 4-13. Detailed view showing toroids.

Westinghouse WC115 or the MC1429G unit can be used, if attention is paid to the different pin connections.

The output keying transistor is a high voltage type, so the standard 150-volt loop supply can be used. Again, almost any type of high voltage transistor can be used, but the Motorola MJE-340 can handle the power easily. The neon bulbs are necessary to protect the MJE-340 from kick-back spikes.

Layout is not at all critical and modifications are easily added for tuning indicators, reversing switches, etc. The best way to put in a reversing switch is to switch the hot end of C1 and C2 between respective places. Wired as shown in Fig. 4-12, the mark frequency is 2125 Hz, with space being 2975 Hz.

In buying the IC's, the Fairchild 710 is recommended since it is less expensive.

With this unit, very narrow shifts are possible. You can copy a 200-Hz audio shift when it is below the noise level and voice communication is impossible. Narrower shifts should present no problem with the proper selection of values for C1 and C2.

Exact shift frequencies are not mandatory for copy since the ratio detector is an FM type of detector. If the received station's tones are a bit off frequency, you probably will not notice any increase in error rate, unless the received signal is very weak or the tones are considerably off frequency. The operation of this unit can be vastly improved when utilizing narrow shift, by the addition of a good audio bandpass filter,

designed for the shift in use, between the receiver audio output and the terminal unit input. If the signal being copied fades below copy level, or there is no signal at all, the Teletype machine will not run open; it will sit quietly until a proper signal is tuned in.

THE TUZ TERMINAL UNIT

This terminal unit is a transistor type. It is about as good as the best and superior to most. In the process of developing it transistors have been added and the circuit complexity increased somewhat, however the results have fully justified these additions.

This terminal unit was designed solely for performance—the only limitation on complexity was that it be easy enough to be readily built by the average ham. In terminal units there are three areas for better performance—first is in the basic electronics of the unit, limiters, adders, triggers, and selector magnet circuitry; second is the area of filters independent of the electronic circuitry; and third is in the retiming and signal processing techniques used in regenerative repeaters. This terminal unit does an excellent job in the first area, while in the second area—filters, it uses good basic design and includes optional plug-in facilities for those who want to build and use the better types of modern filters. In incorporates no facilities for retiming, but the design is such that this may easily be incorporated later if it is desired. At this stage regenerative repeaters are relatively rare in ham TU's, but there is no reason that it could not be added later on, for a transistor unit should not be too difficult to build.

Design

The unit is basically completely electronic and self-contained: no relays, polar or otherwise, are used to cause hash and get out of adjustment, and no separate power supplies are needed. You simply feed the audio tones from the receiver and plug in the selector magnet of the printer. It also provides local copy and concurrently an adjustable FSK bias voltage for use when transmitting. The only external circuitry necessary is the conventional transistor or diode frequency shifting network on your oscillator.

The design of the unit is straight forward. The basic functions are given in the block diagram in Fig. 4-14. The audio input from the receiver is first fed through an effective limiter consisting of two silicon junction diodes, D1 and D2. Since the

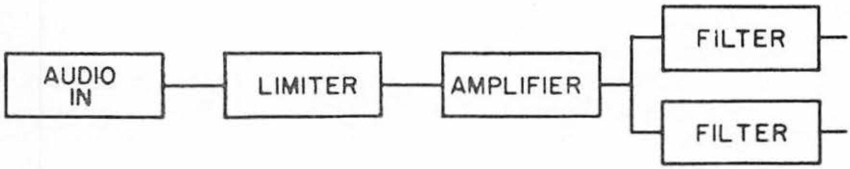
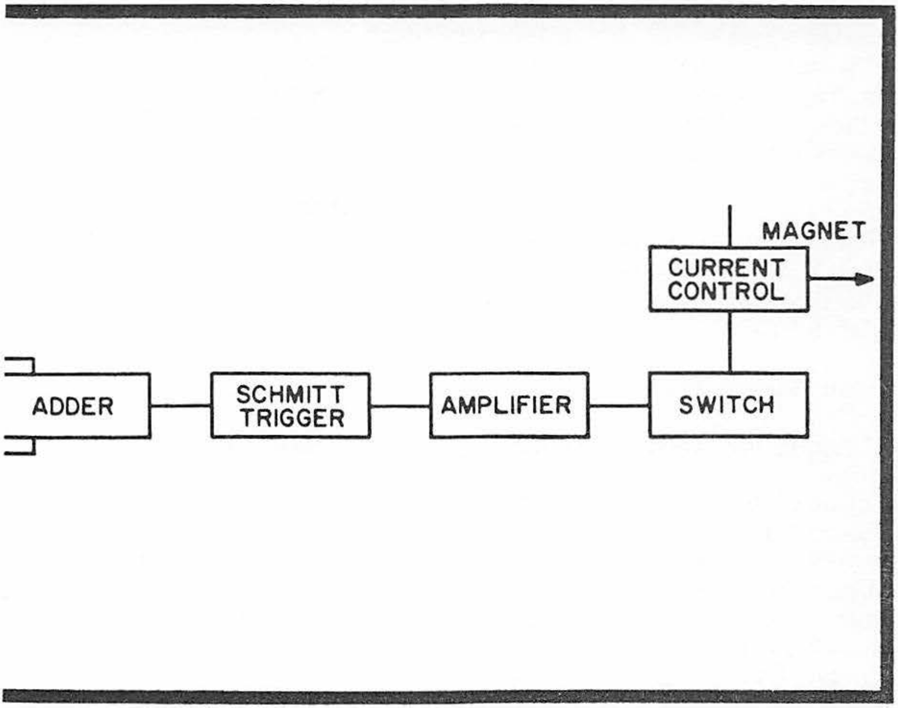


Fig. 4-14. Block diagram of the TU2 terminal unit.

peak voltage across either diode is limited by the forward conduction voltage of the other diode, the PIV requirements are negligible and practically any silicon junction diode is usable. The output of the limiter is then amplified by transistor Q1 and the output is fed to the two filters. The output transformer T1 is an ordinary tube type, 2000 ohms to voice coil. Special transistor transformers are avaiable, but they are usually more expensive and often less efficient, while the saving in space is not significant.

The low impedance output of the transformer is ideal for coupling to the usual toroids used in the filters, since relatively few turns are needed and the number can be easily adjusted to equalize the outputs of the two channels. The DPDT switch following the filters provides for mark-space reversal. This is necessary in cases where the transmitting station has the mark and space frequencies reversed, and is quite handy when you have your bfo on the wrong side of the signal since it is far easier to throw the switch than to retune the receiver. The outputs of the two filters are then rectified by diodes D3 and D4. These are placed so that you have opposite polarity and the outputs are filtered and summed in RC networks. If



you have equal signals in both channels and equal outputs, the summed voltage will be zero—thus you have the usual cancellation characteristics of an FM discriminator. Capacitor C7 and switch SW2 are provided so that you may choose either AC or DC coupling. Briefly stated, in the AC position (SW2 open) the magnet current will be switched on a change in the level of voltage from the discriminator but the resting position may be adjusted so that the machine does not run open when no mark tone is being received. This is quite convenient in eliminating the various misprints one gets when the mark tone is temporarily lost, and also provides protection against the effects of a continuous unwanted signal in either of the two channels. In the DC position (SW2 closed), you have a greater range of adjustment for various types of distortion, although there is still a considerable range of adjustment available with AC coupling. Potentiometer R8 is a sensitivity level control and furnishes protection on occasion against misprints caused by relatively weak unwanted signals. It should be adjusted for optimum copy. When interference is not a problem it should normally be set so that the typical voltage swing at its output is equal to the range of voltage adjustment obtainable from

R14. This permits the widest range of adjustments for distorted signals with maximum sensitivity.

Switch SW3 has three positions: local receive, and transmit. The local and the transmit positions are the same except for the extra set of contacts which are used for station control on transmit. The resistor network R9 and R11 and zener diode D5 give a variable output voltage for controlling the frequency shift of the oscillator of the transmitter and at the same time provide a voltage to the subsequent stages (via resistors R10 and R13) which furnishes local copy. The FSK voltage can also be used for keying an AFSK oscillator if you want to join 2-meter RTTY.

Circuit Description

Transistors Q2 and Q3 form a complementary emitter follower, providing both isolation and impedance transformation, the circuit being analogous to the conventional vacuum tube cathode follower. The complementary circuitry provides for equally effective operation on both rising and falling waveforms. Transistors Q4 and Q5 form a Schmitt trigger and the output from the collector of Q5 is a square wave—regardless of the shape of the input signal. Potentiometer R14 provides an adjustable trigger level for this circuit and permits a wide range of adjustment for distortion of the incoming signal. Transistor Q6 performs three functions: it serves as a buffer-amplifier (mostly buffer) between the Schmitt trigger and the switching transistor (Q7), it provides the phase reversal which is necessary for using the FSK voltage for local copy, and it permits the application of a positive bias to the switching transistor to assure effective cutoff of the transistor.

Transistor Q7 is simply a switch to turn the selector magnet current off and on. Since the input is a square wave (i.e., either off or on) the actual power dissipation is very low. The steady state dissipation is 100 milliwatts or less in either the off or on state, and during switching it reaches an instantaneous peak value of about 1 watt. The average power dissipation on continuous reversals (RYRYRYRYRY.....) should not exceed 150 milliwatts. Actual transistors tested have ranged from 300 mw units up to the largest of power transistors—in practice medium power units are recommended merely because of the added safety factor. The current regulator transistor Q8 and the associated circuitry provide a very low resistance path when the magnet current is less than the required value—when the current reaches the required value, the voltage drop across the transistor in-

creases until it is sufficiently large to assure this desired current. This transistor has to dissipate about 4 watts with current on, thus it must be a medium power transistor with a moderate heat sink. The zener diode D10 provides the reference voltage for this current regulator circuit and a 3- or 4-volt unit is recommended—theoretically the lower the voltage the better, but the actual value is not critical. Higher voltage units may be used with a very slight decrease in performance, but if they are, the value of R28 should then be increased somewhat to protect against excessive current.

Power Supplies

The power supplies are conventional in every respect. The diodes used in the high voltage supply are 100 volt PIV units with a current rating of .25 or more. The bias supply used surplus 1N482 diodes which were available from the junk box, but any diodes rated for 15v PIV, 10 ma or better can be used—and those are very easy specifications to meet. You could probably save some space, components, and initial expense by using a standard 9-volt transistor battery and eliminating the bias supply and zener diode D7. Another possibility which might be tried is to use a simple half wave rectifier in the bias power supply. The current drain here is very low and the filter used should be ample.

With the exception of the diode D10 in the current regulator circuit, all the zener diodes specified are Hoffman HB-1. These are listed as general purpose diodes guaranteed to have a PIV greater than 7.5 volts. They practically all have zener points within the range from 7.8 to 15 volts with the vast majority falling within the 8 to 12 volt range. These are undoubtedly fallout from their regular zener diode production which do not meet their commercial standards—but they are quite adequate for a variety of amateur applications. They are more effective than electrolytic capacitors, since their impedance is not frequency dependent, and in a terminal unit we are dealing with some very low frequencies. In addition they take up a lot less space than electrolytics and as long as you stick to the HB-1 variety they are cheaper. Checking them is a simple matter, just use the test arrangement given in Fig. 4-15. The power supply can be that of the unit. The value of the potentiometer is not critical but the fixed resistor in series with the diode should be sufficiently large to limit the current to less than 10 ma. Just connect the diode and run the potentiometer slowly up. The voltage should rise smoothly until the zener voltage is reached. Increasing the pot further should not cause any significant increase in voltage. If you have less than

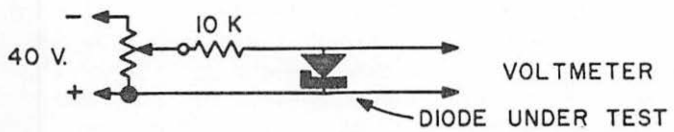
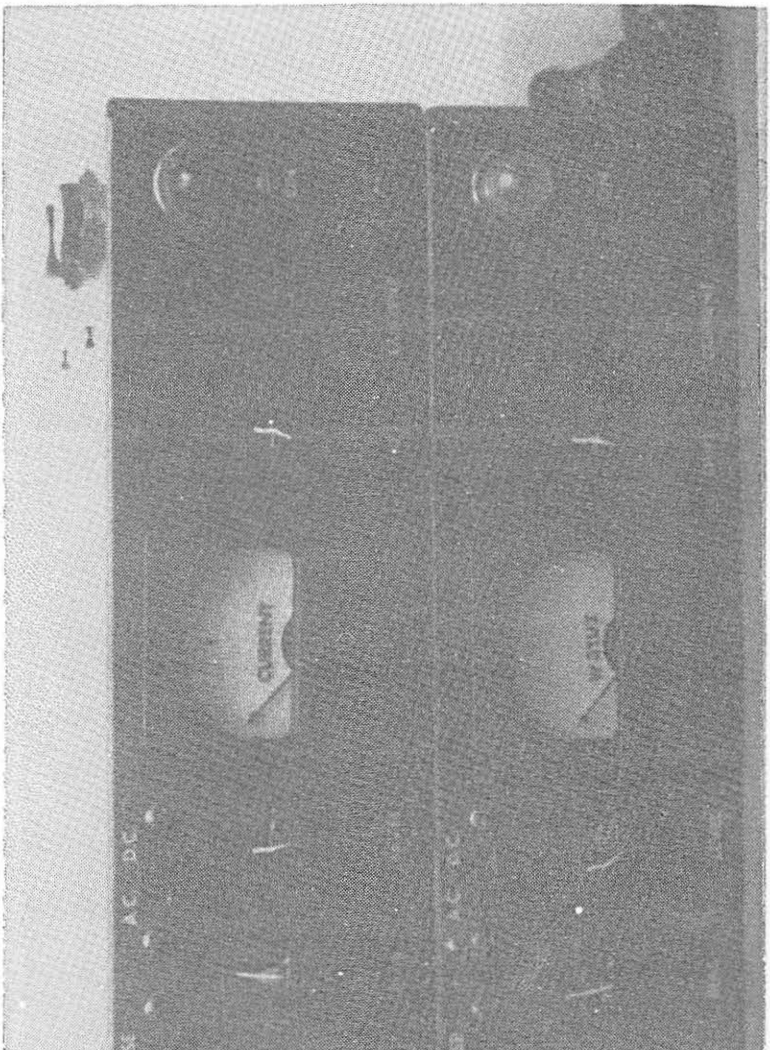


Fig. 4-15. Diode test setup.

a volt or so across the diode, you have the polarity of the diode reversed; turn it around. If you get a continuously increasing voltage with no zener point, either it has no zener point or it is above the voltage range you have tested it for, but in either case don't despair, the diode will be quite satisfactory for either D1, D2, or D6. Test them all at the same time and mark the zener voltage on each. Select one of the higher voltage units, on the order of 10 or 12 volts for D5. Use the lowest voltage unit, preferably between 8 and 9 volts for D7. If you are unable to find one this low, D7 may be replaced by a 100 mfd 12v electrolytic capacitor. As for D8 and D9, the actual voltage is not critical and can be anywhere in the 8 to 12 volt region—however D8 should have a higher zener voltage than D9; a half volt differential is quite adequate. This differential assures that transistor Q6 cuts off.

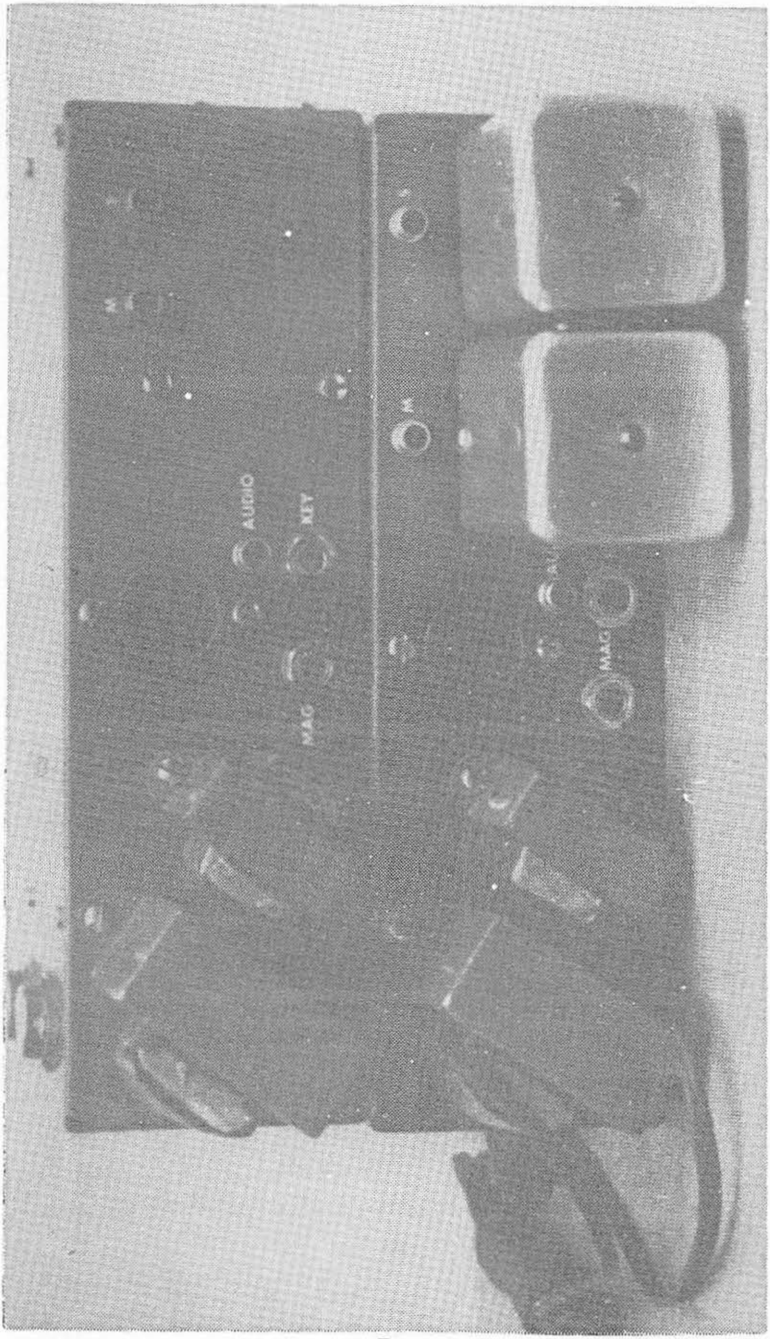
Transistors

The transistor type specified are modern units that you can buy at reasonable prices, plug in, and be assured of satisfactory performance. There are cheaper (but very little cheaper) transistors which can be used if you are willing and able to go through a selection procedure to find the satisfactory units. The 2N2374 and 2N2375 units are low cost modern PNP units with relatively high voltage capability (-35v). They also have high beta without having the exceptionally wide beta spread of some of the less expensive units. If you have some means of checking beta, using the highest beta transistor for Q4 and the next highest beta unit for Q5. The 2N1304 are moderately priced NPN transistors and perform very nicely in the circuit. The output transistors used in the various units have been of a wide variety of types, in general the 2N251 or any T0-3 (diamond base) transistor with a 50-volt rating and



A

Fig. 4-16. Two similar variations of the completed TU2 TU's. (Continued on pages 88 and 89)



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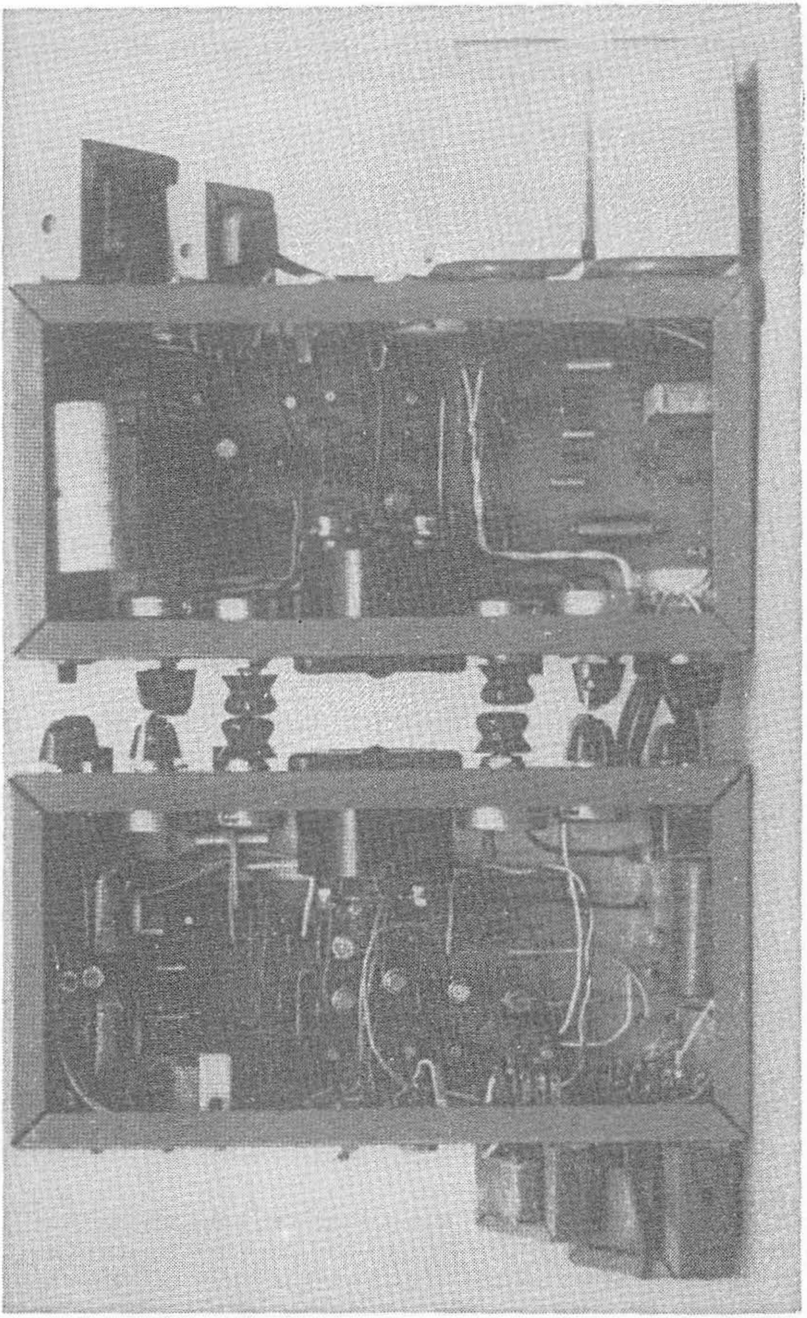


C

reasonable beta will be easy to mount and should do an acceptable job. For those who prefer the stud mount, the 2N1501 has done a fine job in a number of these units.

The Completed Unit

There are several ways that the terminal unit can be packaged. Two similar units illustrating one version are shown in Fig. 4-16. The circuit diagram is given in Fig. 4-17. The units use a small printed circuit board—roughly 3 inches by 8 inches in size. The layout of the board is shown in Fig. 4-18 and the parts layout on the top of the board is shown in Fig. 4-19. This board provides a large percentage of the circuitry of the unit, but since it provides for separate filters, controls, output transistors and power supply, it does provide a considerable degree of flexibility in packaging. It can, of course, be readily adapted to relay rack mounting by the use of a standard 3¾ inch relay rack panel. One of the units pictured uses 2N251 transistors and has the toroid filters mounted integrally in the unit. The other unit shown uses plug-in filters and has 2N1501 transistors mounted internally. This actually has an extra transistor mounted internally to be used as an additional output for a reperforator. This permits a reperforator to be used along with the page printer and gives separate control of magnet current. The circuitry for this modification is shown in Fig. 4-20. The jack which is paralleled with the magnet output of the octal plug on the other unit is here wired up to handle the extra output. The plug-in filters may be mounted in the Vector C-12 cans shown in the photo, or may be more complex units mounted in a 3" x 5" x 10" chassis—which is the chassis size used for these two units.

With this size circuit board you can choose your own size cabinet and your own panel layout—the space occupied by the board is small indeed. Please don't get the idea that this represents the ultimate in packaging techniques and shrinkage. If you want a smaller unit, just use smaller size potentiometers, wind a single transformer for the two voltages required, use a miniature meter and miniature electrolytics.

Filters

The filters used in the terminal unit are bandpass units of the conventional type. The circuits and the component values are given in Fig. 4-21. There is also an illustration of the set up for tuning the filters. Tuning is simple using the test set up. Connect it to the LC pair to be tuned, short out the other LC

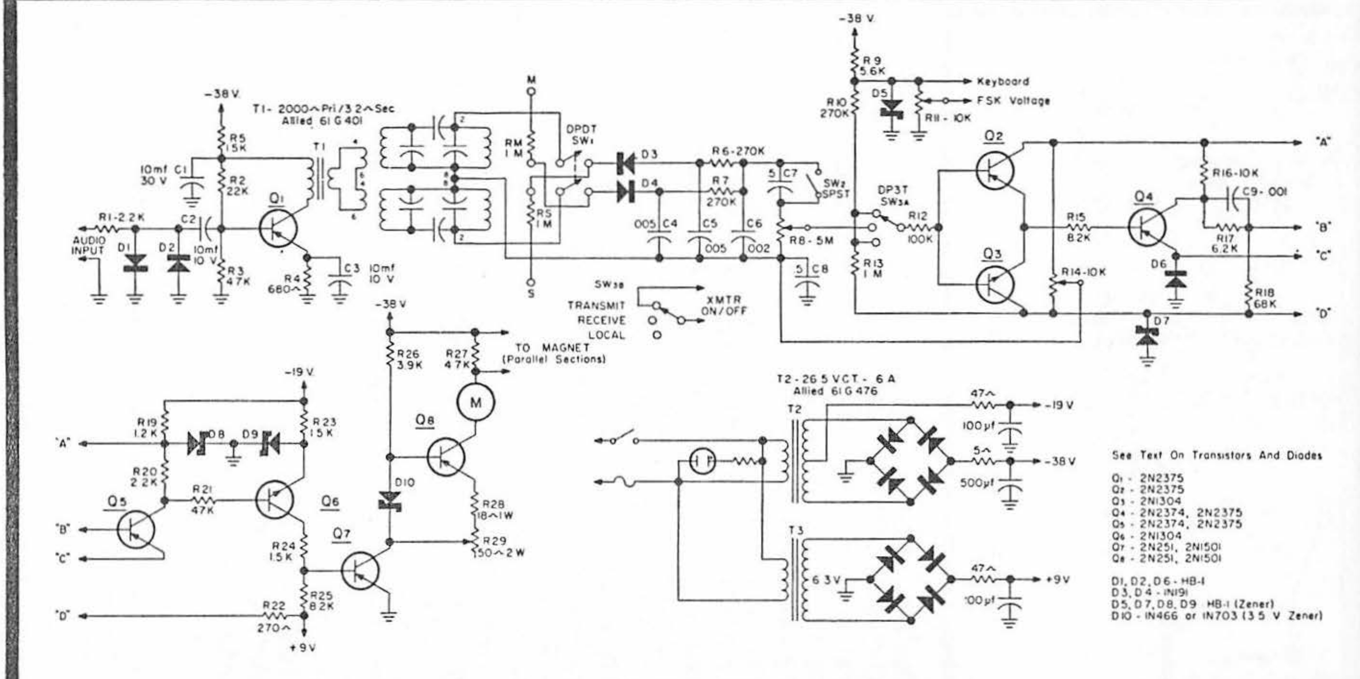


Fig. 4-17. Schematic diagram of the TU2 TU.

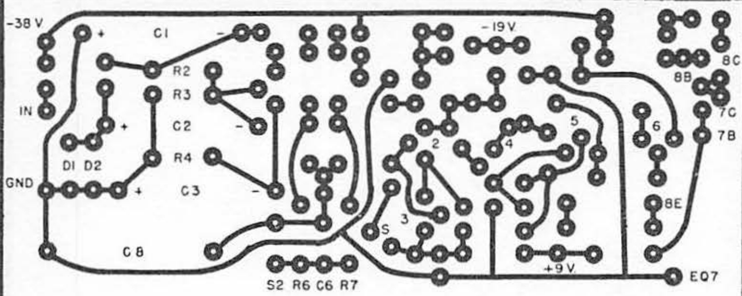
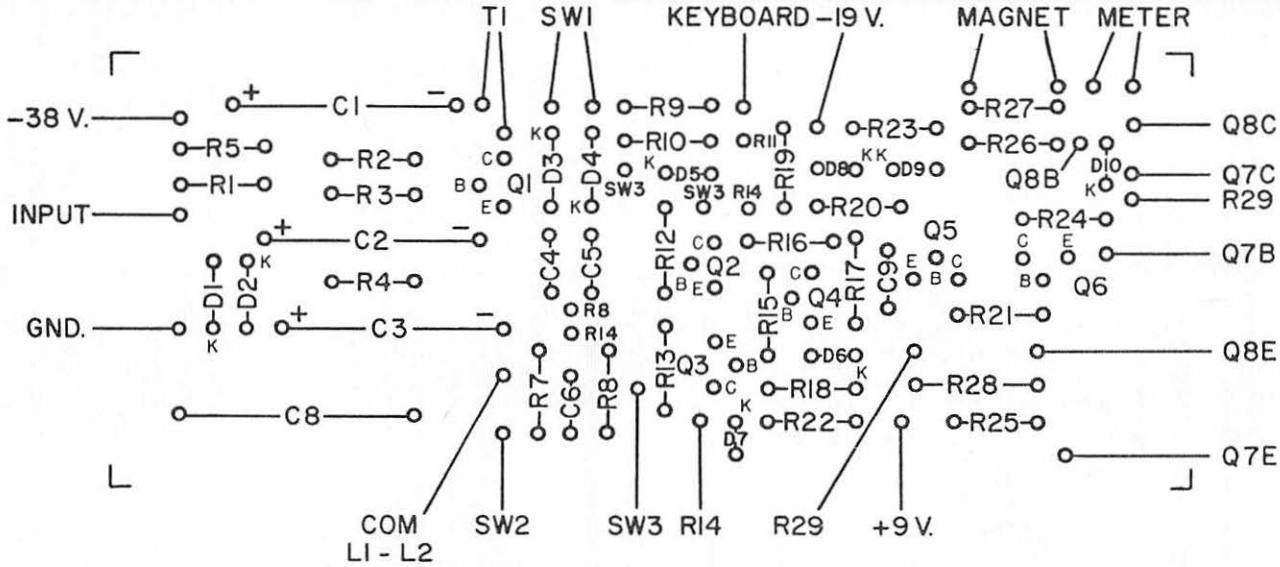


Fig. 4-18. PC side of board.

pair of the filter you are tuning, and either by trying different capacitors, or by adding or subtracting turns from the toroid, bring that section to resonance on the desired frequency (either 2125 or 2975 cycles). Then switch the short to the section you have tuned up and proceed to tune the other section. Note that this procedure effectively adds the coupling capacitor to each LC pair. The coupling capacitor is chosen from normal 10 percent capacitors, which should be close enough for proper performance. You should preferably tune the output section with the load connected so that tuning does not shift after the unit is wired up. The values given are for a bandpass of about 200 cycles, which is our recommendation for general use. If toroids or space are problems, you may prefer to use only one toroid in each section; this will not give quite as good performance as the bandpass unit with two toroids per section, but the performance will still be considerably better than the average terminal unit. Just omit the coupling capacitor and the second LC pair.

When building the unit you can either mount the filters on a terminal board and put it inside the unit, or you can make each of the filters a plug-in unit. The numbers at the terminals are the base connections for the octal sockets used with the plug-in units. It is handy to have these standardized, to check out terminal units before the filters are finished. The plug-in units also permit you to have other filters with narrower bandpass or special filters for short shift. By making up just one filter for 2550 cycles, you are equipped for copying the commercials using 425 cycle shift. The dual plug-in units are each mounted in a Vector C-12 can which is 2" x 2" x 3" in size. The Millen 74400 can may also be used. When the units are finished and connected in the circuit, check to see that the



NOTE: C7 Is Mounted On Switch SW2 On Front Control Panel.
K Is Marked For Cathodes Of All Diodes.

Fig. 4-19. Component side of board.

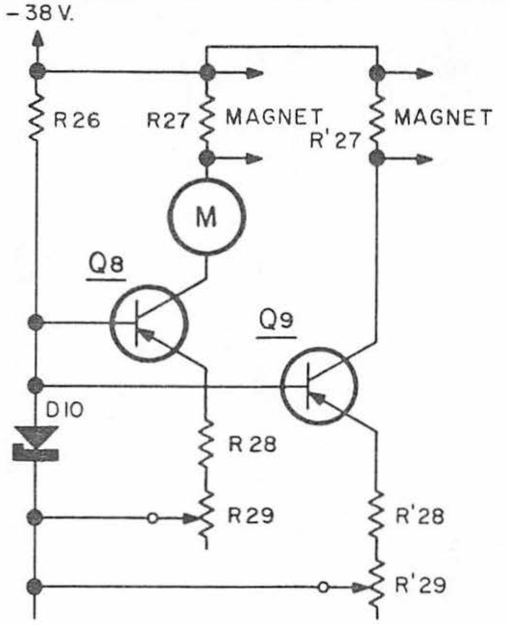


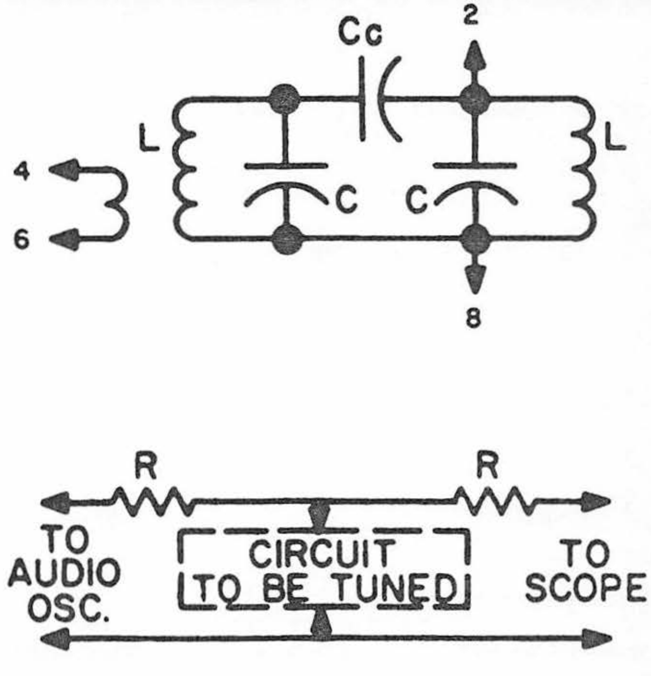
Fig. 4-20. Circuit for operating two printers.

bandpass is smooth across the top and not double humped. If you do have a double hump with the units operating with the normal load, add a resistance across the output toroid of each filter. By careful selection of the resistor you should have no difficulty in smoothing the response. Without a load you will surely have a double humped curve, and the value of resistance needed will depend to some extent on the betas of transistors Q2 and Q3.

Testing the Unit

Checking out the unit is a relatively simple procedure. For the initial check, disconnect the lead supplying voltage to Q7 and Q8. Plug in the unit, apply power, and check the power supply voltages. First check the voltage at the collector of transistor Q5. By varying the setting of R14, the voltage should at some point shift from a little less than one volt to approximately the zener voltage of D8. You should not, repeat not, be able to obtain an intermediate voltage reading regardless of the setting of R14. If you can, the transistors you

are using for Q4 and Q5 probably have two low a beta. The switching should occur when R14 is set approximately to center scale, i.e., when the voltage on the center arm of the pot is about -1 volt. One unit, on which detailed voltage measurements were made switched on a .03 volt change on the bases of Q2 and Q3. The next step is to check Q6. The collector voltage should switch from about the zener voltage of D9 to about zero volts. A further check should be made on the voltage from the collector to the emitter of Q6. When this transistor is conducting, the voltage drop should be less than one volt. If it is higher than this, either use a higher beta transistor for Q6 or decrease the resistance of R21.



L all 88 mh		R = 1 meg. or larger
	2125 cycles	2975 cycles
C	.06 mfd.	.03 mfd.
C_c	.0059 mfd.	.0022 mfd.
Link	9 turns	5 turns

Fig. 4-21. Filter tuning setup.

Next check the mounting of transistors Q7 and Q8 to see that you have not grounded the collectors in the process of mounting and heat sinking them. After reassuring yourself on this point, connect the power to the units, plug in the selector magnet, and turn on the unit. By now, varying the setting of R14 you should be able to switch the magnet current off and on. Then, with the current on, adjust R29 for the proper magnet current. For Model 14 or Model 15 machines with pulling magnets, be sure the two sections of the magnets are in parallel and run 120 ma to the paralleled magnet sections. With holding type magnets, less current should be satisfactory. If you are unable to draw the full 120 ma with R29 set at minimum resistance, you probably have a very low zener voltage diode for D10 and the cure is simply—merely decrease the resistance of R28. Normally, you set R29 and leave it alone, so if space is a problem you can locate this pot at any position on the chassis you have the room for it. Alternatively, you could merely select R28 for the proper current and delete R29 completely. Check the voltage from the collector of Q7 to ground. When the magnet current is on this voltage should be less than one volt. Readings of .1 volt to .4 volt are typical. If it is more than a volt, the beta of the transistor is probably low—but with identical transistors being used you can easily switch transistors and see if this improves the situation. Actually when you use power transistors here you are normally well off since the betas are usually considerably higher at the 120 ma operating point than they are at the full rated current the devices are designed for. You can compensate for an extremely low beta unit by decreasing the resistance of R24 from 1500 to 1200 or even 1000 ohms. A number of the units built have used 2200 and 2700 ohms for this resistor, but the 1500 has been specified to provide latitude for the lower beta transistors.

Assuming all works well, the only remaining step is to feed audio tones into the unit and see if it prints. The audio tone is limited by D1 and D2 (an oscilloscope will demonstrate this visually), it is then amplified by Q1, and the two filter sections separate the tones. With a mark tone input you should get a positive output at the junction of R6 and R7, while for a space tone you should have a negative output. If they are reversed, just throw SW1 and the situation should be as specified. With SW3 in the receive position (center) these tones should switch the magnet current on and off.

A teletype terminal unit that is better than the majority of the TU's presently being used by amateurs on RTTY is called the Chemical City TU (Fig. 4-22), after the "chemical city" of

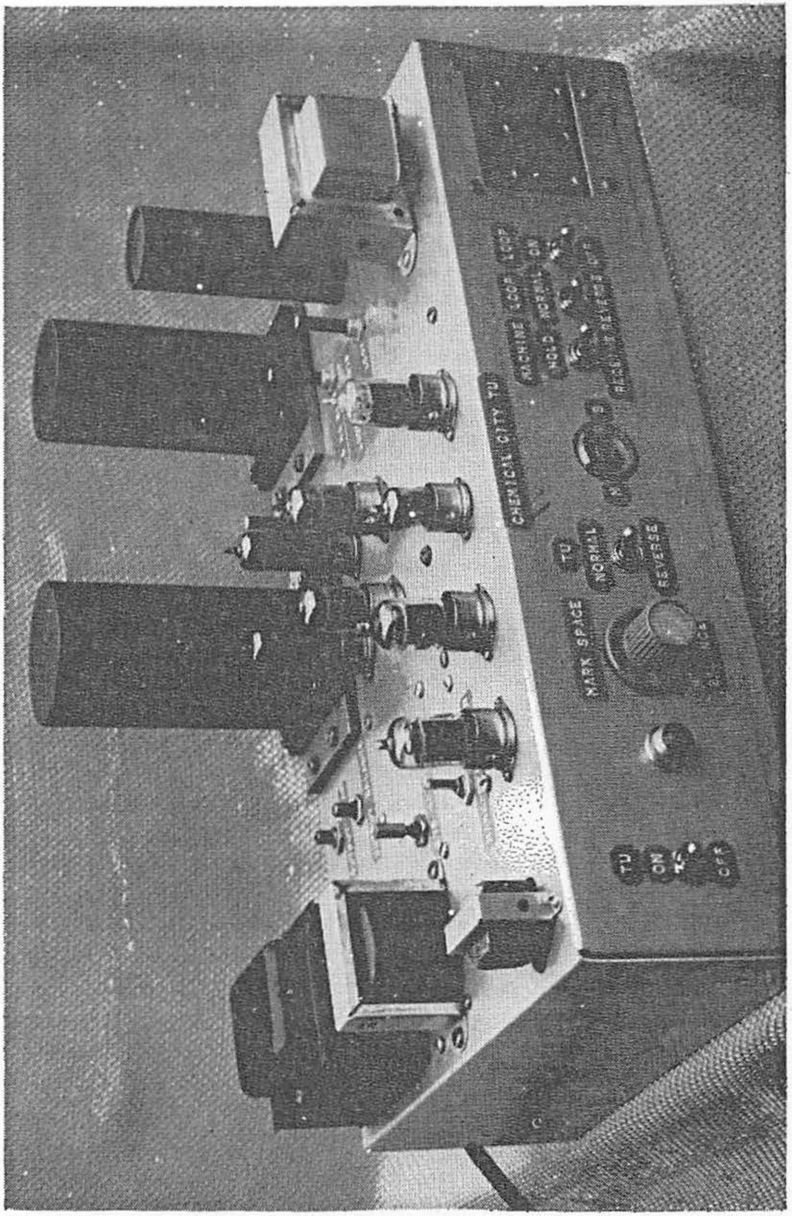


Fig. 4-22. Chemical City Terminal Unit.

Midland, Michigan. It features provisions for copying on either mark or space or both. It also permits copy of signals too low in the noise to be usable printable material except by the very best terminal units, which are usually not available to the average amateur.

Circuit Description

One of the unique features of this TU appears at the secondary of the input transformer (Fig. 4-23). As can be seen, it utilizes a pair of 1N2701 silicon diodes for the first stage of limiting. These diodes, because they have the property of passing one-fourth of a volt before they start to conduct, are ideal for limiting purposes without the use of bias voltage and the associated resistors necessary with the conventional diode limiter. The use of these diodes results in a total of ½-volt peak-to-peak on the grid of the 12AX7 second limiter amplifier. With this arrangement, the signal output from the 12AX7 is constant between the level where the signal is just audible to the ear to full output from the receiver.

The signal voltages then pass through a 12AU7 amplifying stage; one triode section for the mark channel and one triode section for the space channel. In the plate circuit one will note that the pair of transformers drive the 6AL5 balanced detector. The output of the balanced detector is used to feed three different tubes in the converter as can be seen in the schematic.

The first tube fed by the detector is another 12AU7 which is a DC amplifier and keyer for the mark and space coupling. Output from the detector also feeds the 12AU7 preceeding the detector. And finally, the detector voltage is fed to a 12AX7 inverter tube, which begins the process of converting the trailing edges of mark pulses to space pulses and space pulses to mark pulses making it possible for one to copy on either mark or space when one of the two signals is not present for reception.

A second unusual idea is the feedback loop which is associated with the application of voltage, derived from the output of the 6AL5, on the grids of the first 12AU7 amplifier. This voltage causes three things to take place. First, the DC component from the output of the 6AL5 provides bias on the grid of the 12AU7 giving the converter a small amount of automatic gain control. This gain control begins operating when the signal into the limiter is of lower amplitude than the operating point of the limiter, at which time, the automatic gain control takes over to hold the signal through the con-

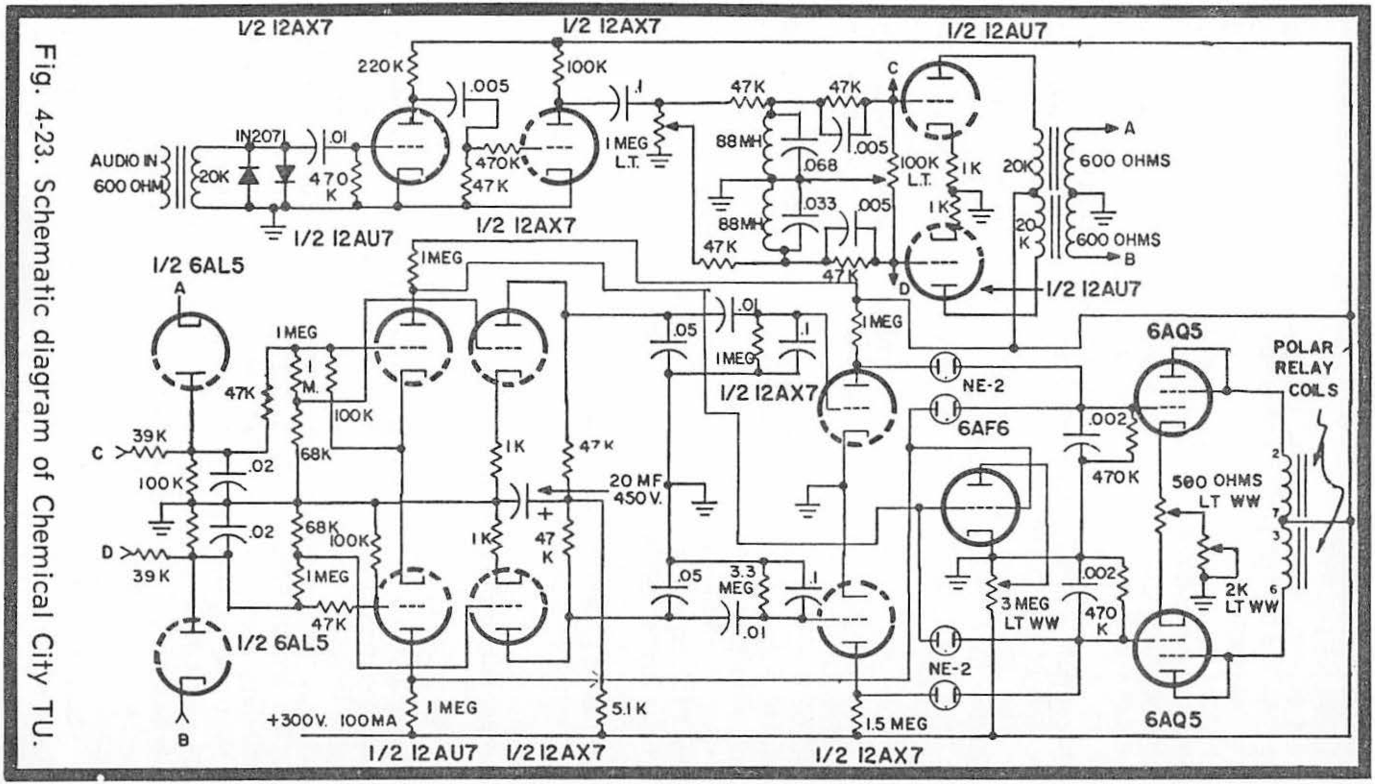


Fig. 4-23. Schematic diagram of Chemical City TU.

verter at a more constant level than would normally be had if it were not present.

A second benefit derived from the voltage on the grid of the 12AU7 is that of giving pre-emphasis to the pulses as they pass through this stage before going on to the 6AL5 for detection. A close look at the operation of the detector will show that the voltage at the output of the detector is a function of the signal at any given time. When the signal changes from mark to space and back to mark again, the bias on the grid of the 12AU7 amplifying tube changes with respect to the incoming pulses. This action produces a more definite pulse at the grid of the 12AU7.

The DC output from the detector, which is actually negative half cycles of the audio going into the detector, is utilized in this converter, as in other converters, to operate the various keying stages. However, along with this pulsating DC voltage, there appear negative pulses of noise which will trigger the keying stages, giving erratic operation on marginal signals. Feeding these negative pulses back into the grid of the first amplifying stage tends to cancel out the positive pulses of noise appearing at the grid, reducing the noise figure of the system.

A 6AF6 dual-eye tube was incorporated in the TU to aid in tuning the signal in correctly. The grids of the 6AF6 are fed directly from the first 12AU7 keyer tube. The 3-megohm potentiometer is used as a voltage divider in the plate circuit of the 6AF6 to set the closure of the shadows to a hair line when the limiter is saturated.

Component Arrangement

The parts layout is not critical except that the input of the TU should not be too close to the output of the 12AU7 amplifier as it may tend to oscillate. The only critical part needed in duplicating this converter would be the type of transformers used to feed the detector. The original model utilized a pair of surplus transformers with a 20K primary and a 600-ohm secondary. Almost any similar transformer could be made to work; however, the 47K resistors at the output of the detector, which feed the 12AU7 keyer, would have to be changed and the voltage dividers feeding the 12AX7 inverter tube would also have to be changed. Otherwise, one should have trouble duplicating the unit.

There are other improvements that could be made on this circuit which would make the converter even better. The addition of a comb filter at the input would further reduce the

noise entering the TU. Also, bandpass filters at the input could be used for greater suppression of unwanted signals and for greater ease in tuning. The addition of an axis restorer to hold the machine in absence of any signal; and the addition of a mark-space axis computer, to move the operating point of the detector half-way between the mark to space peak amplitudes, would also improve performance.

The Chemical City TU shown also contains an integrated loop supply on the same chassis. Fig. 4-24 illustrates that there is room available for future modifications.

AN ADVENTURE IN DESIGNING A TU

The Model 15 machine requires 60 ma through the selector magnet to operate. There are two sections to this magnet; and, with the sections in series, the resistance is about 200 ohms and the required current is 60 ma, or one could connect the sections in parallel (50 ohms) and use 120 ma. Either 12v at 60 ma or 6v at 120 ma (total power of .72 watts) should do the trick! Alas, would that it were true. It is a longish tale, but read on and if all goes well, you may be spared some of the trauma one RTTY'er experienced.

The next step is to acquire a terminal unit. Two audio tones in it are used to turn the magnet current off and on. The keying tube is a 6L6, with more than 250 volts for the B supply. Here is almost 6 watts of heater power and over 15 watts of plate power used to deliver the .72 watts to the magnet.

The Search for a Better Mousetrap

There is an easier way to do this. The obvious approach is to use transistors. As current operated devices, they should indeed be fine for switching the magnet current off and on. In other portions of the circuit they should be equally good. The highest frequency in the unit was 2975 Hz and most of the circuitry would be dealing with 22 millisecond pulses—certainly no fancy transistors would be needed for those frequencies. So a "Mark I" transistor terminal unit was built, by merely substituting a transistor stage for each tube stage in the circuit (which, in general, is the easiest but not the best way to design equipment). It worked, but the performance was less than sparkling. The next step was to design a unit following a block diagram of the functions desired; the result was a complete FSK unit using only three transistors. It worked better, and in fact, a fair number of copies of this unit have been built. It performed fairly—but, not as well as the

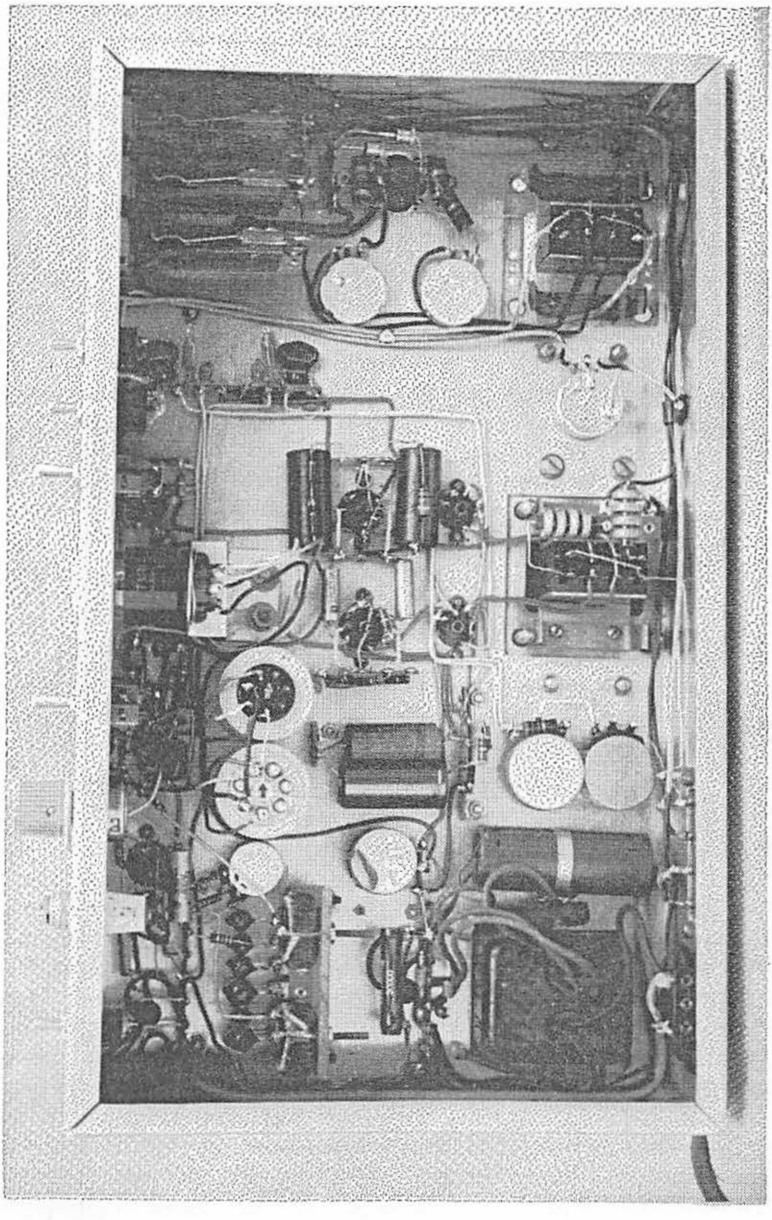


Fig. 4-24. Interior parts arrangement.

tube unit. The feeling persisted, however, that it should be feasible to build a transistor unit that would perform as well as the tube unit, and do it in a simpler, more compact, and more efficient manner.

Watching Waveforms

For research, hook up a scope to monitor current and feed in 22 millisecond pulses. This is easily accomplished, by putting a 1 ohm resistor in series with the selector magnet of the machine and connecting the scope input across this resistor; the voltage across the resistor will be a function of the current through the selector magnet and the extra resistance will have a negligible effect on the operation. (See Fig. 4-25A.) One word of caution, a DC scope is an absolute necessity. The frequencies involved are low enough so that practically all AC coupled scopes will distort the wave shape. If there is any question in your mind on this point, use a relay, a battery, and a resistor. Key the relay at the pulse rate, connect the scope across the resistor, and observe the wave shape: if it is not square, it is the fault of the scope. (See Fig. 4-25B.) Incidentally, if you have an electronic key, it is a fine square wave generator; just set it for dots at about 22 or 23 dots per second. Alternatively, 4 or 6 volts AC into a polar relay will do the job.

If the traces are not square waves, then the root of the problem is the selector magnet. We are applying a voltage to a device which consists of a resistance and an inductance. If the inductance is fixed, you get the smooth curve in Fig. 4-26. However, the inductance in a teleprinter is not a fixed value. When the armature is not against the pole piece, the inductance is about 2 henries; when current is applied, the

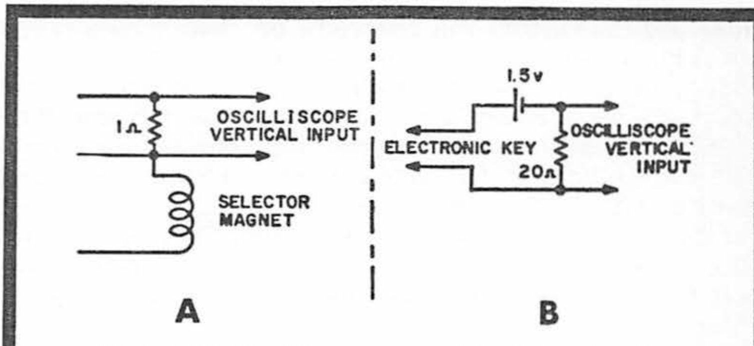


Fig. 4-25. Oscilloscope vertical input setup.

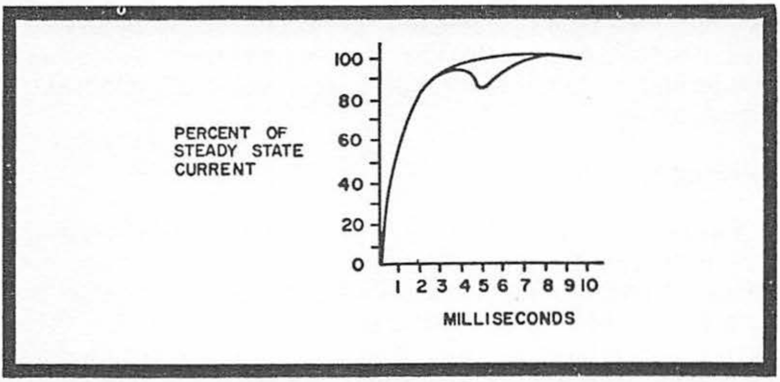


Fig. 4-26. RL current rise curve showing jog where transfer from mark to space occurs.

magnet charges, and as the field increases, the armature is drawn in. The physical movement of the armature is accompanied by an increase in the inductance. For purposes of analysis, this is convenient, since this change in inductance causes a jog in the current curve, and gives a visual indication on the scope of the point at which the mechanical transfer from mark to space occurs (see Fig. 4-26). All the wave shapes here have a time base on the horizontal axis and selector magnet current on the vertical axis.

As a starting point, consider the basic land line printer circuit for the Model 15, with a supply voltage of 120 volts, selector magnet sections series connected, and a resistance of approximately 1800 ohms to limit the current to 60 ma. The time constant of the circuit in milliseconds is the inductance in henries divided by the resistance in ohms (the limiting resistance plus the resistance of the selector magnet). In this case the time constant is 1 millisecond, which means that in 1 millisecond the current will rise to 63 percent of its steady state value, in 2 milliseconds it will rise to 86 percent of its steady state value, and in 3 milliseconds it will rise to 95 percent. By using even higher voltage we can increase the series and thus achieve a still shorter time constant, however, we have a mechanical motion of the armature involved, and, beyond a certain point, an even shorter electrical time constant will not give significantly faster mechanical movement. Hence, any efforts in this case will result only in an increase in power comsumption without any corresponding gain in performance.

What happens mechanically under these conditions is that there is a delay of about 3 milliseconds between the closing of

the circuit and the mechanical transfer; and, at the end of the pulse, if the spring tension is properly adjusted, there is also a 3-millisecond delay in the return to the resting position. This adjustment gives the maximum range for the machine. The effects of less than optimum magnet currents are obvious from the family of curves in Fig. 4-27. At lower current levels the delay in the mechanical transfer becomes progressively longer and the action slower. In fact, it is possible to have a current level such that the delay equals the pulse length. The result is that the range of the machine becomes progressively smaller and the operation correspondingly less reliable. It might be noted that for the relatively slight decreases, it is possible to adjust the spring tension, and thus the delay or drop out to compensate for the increases initial delay. However, for optimum performance, it is far better to accept the manufacturer's word on the optimum current value.

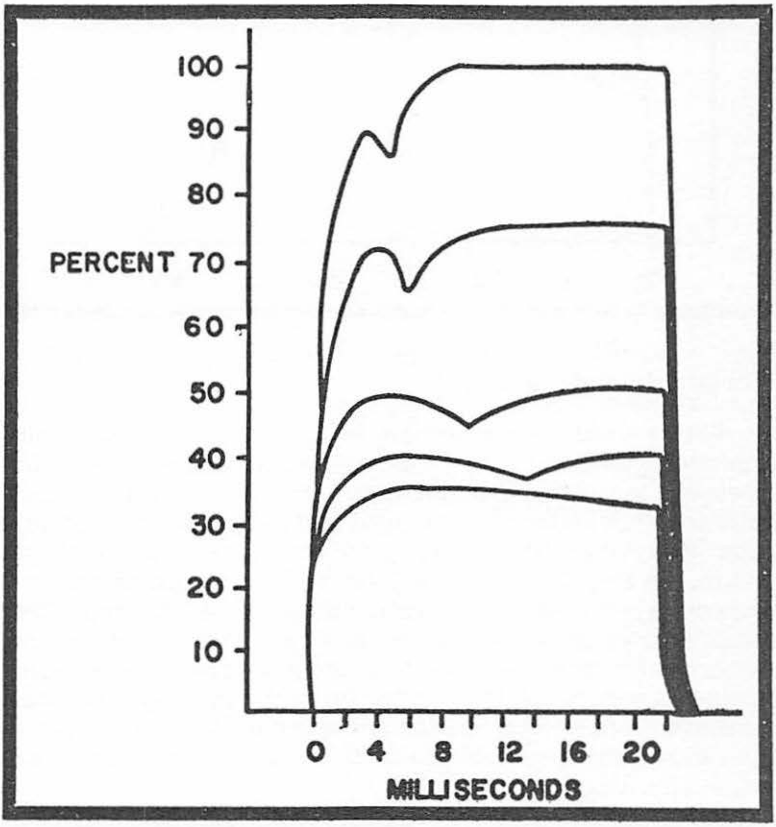


Fig. 4-27. RL current rise curves showing effects of varying the adjustment.

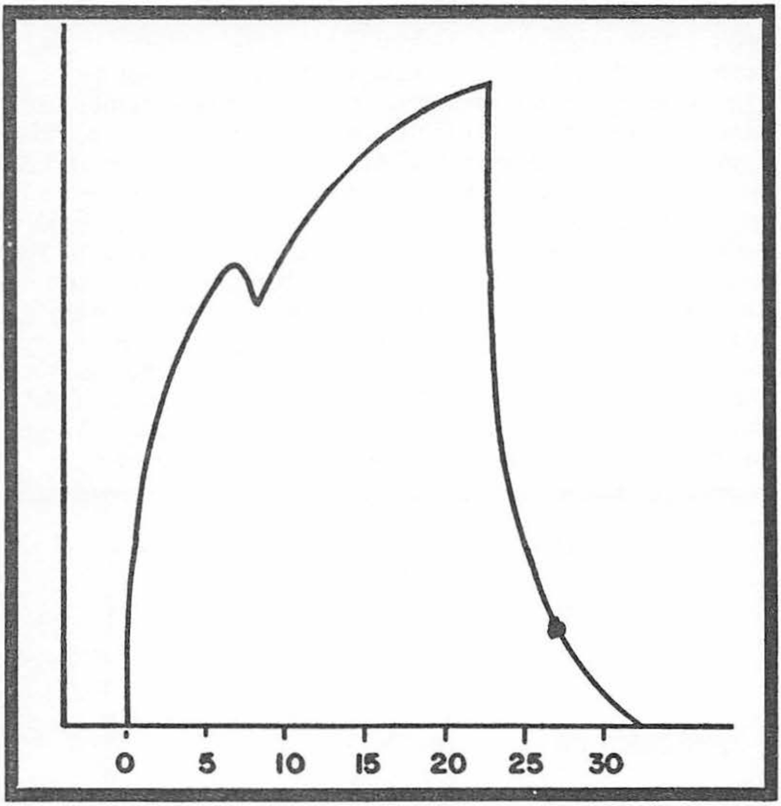


Fig. 4-28. RL current rise curve of a commercially marketed solid-state unit.

With regard to this process of lengthening the drop out time to compensate for slow rise time, we ran across an interesting commercial application. One of the commercial units that is marketed as a solid state replacement for the polar relay was tested and gave the current wave shape pictured in Fig. 4-28. The rise time is slow but the circuit has been designed to give a correspondingly slow fall time. The result is that the range of the machine remains about the same. With a machine which is not properly adjusted, this might even increase the range. In on-the-air tests this was more sensitive to interference and gave poorer performance than either the best tube units or the transistor circuit we eventually used.

In a typical electron tube terminal unit that provides magnet current through a keying tube, we normally have 250 volts or more and a tube such as a 6L6 or 6Y6, which can

handle the current satisfactorily. The variable series resistance and the plate resistance perform the current limiting function. The time constant is even shorter than that of the manufacturer's recommended circuit, and if properly driven by a square wave input, the performance is indeed good. A tube having a sufficiently high perveance should perform well with a 120-volt supply, although no tests have been made on such units.

Next, consider the case of a transistor. If we merely replace the switch or relay in the conventional keying circuit with a transistor, the performance should be practically the same as the manufacturer's recommended circuit. The transistor, having a far lower saturation resistance than a tube should do better than a tube. If you start pricing transistors that will operate comfortably at 120 volts, you immediately conclude that theory and pocketbook don't always agree. In addition, the other transistors in the unit don't need voltages of this order, so you would either have large dropping resistors or need two separate power supplies. Neither alternative is particularly appealing.

An example of this approach uses a grounded-base high voltage transistor (this helps on the voltage capability side, but makes driving the device far more difficult) and requires three different power sources. Another effort uses three 45-volt transistors in a complex series arrangement. Neither of these alternatives are particularly attractive. The next step is to see what will happen with a pedestrian transistor and a 40-volt supply. The discouraging results are reported in Fig. 4-29. This will give copy, but it will not give good range or compete with the better tube terminal units.

The next step along the path is ridiculously simple. If we use 60 ma through each section of the magnet for satisfactory operation, we can do it with 60 ma and the sections in series, or 120 ma and the sections in parallel. Now when we put the sections in parallel, we have .5h inductance rather than 2 henries. Consequently, with 60 volts at 120 ma we have the same time constant and the same performance that we had with 120 volts and 60 ma. The increased current is easily handled by a transistor, whereas a tube to do this would be a fairly substantial bottle. At this point the voltage requirement has been cut by a factor of two, but 60 volts is still too high for convenience; i.e., a reasonably priced transistor. We really want something that will operate in the 30- to 40-volt region (with a bridge rectifier circuit and 15 to 20 volts available for the signal processing circuitry) and will yield all the performance of which the complex machinery is capable.

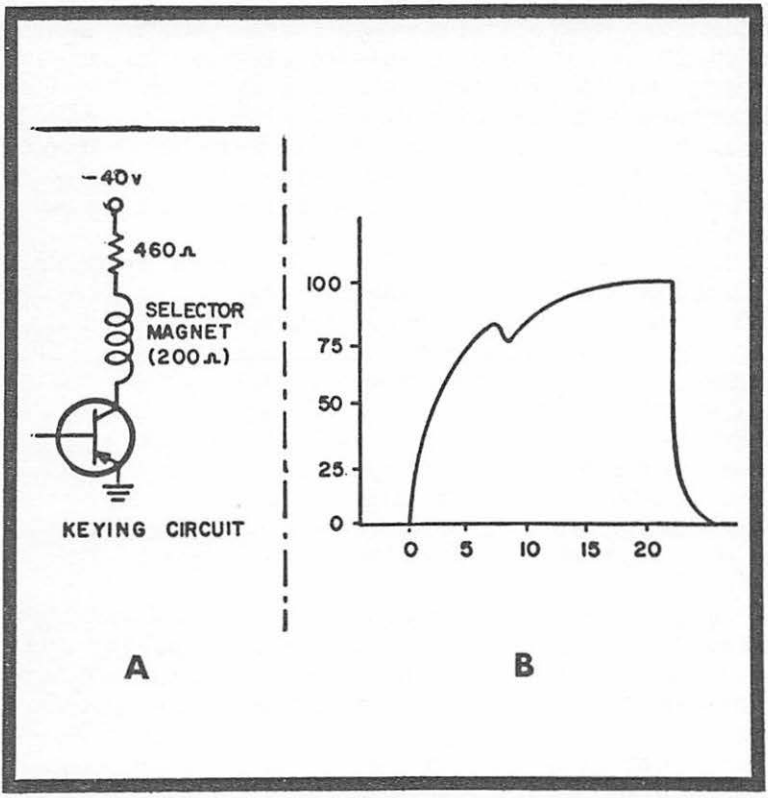


Fig. 4-29. Circuit and resulting unsatisfactory curve using pedestrian transistor.

The next step is a closer look at the basic keying circuit. (Fig. 4-29A) This is not really what is desired. Better performance would be possible if the resistance could be made a direct short while the magnet is charging, and when the current reaches the desired level, the resistance increases in value to that necessary to limit the current. It not only has to do this, but should be sufficiently fast acting so that an added time constant is not super-imposed on the one we are struggling to reduce.

To do this you use a current limiting circuit which uses a transistor, a zener diode, and two resistors (Fig. 4-30). In this circuit R½ is chosen so that the voltage drop across it with full operating current is equal to the zener voltage of D1. For a 3.5-volt zener diode this required a 30-ohm resistor. When the current is less than the required current, transistor Q1 is in effect a very low resistance; when the current reaches the

desired value, the effective transistor resistance increases and the voltage drop across the transistor increases to that necessary to maintain the desired voltage.

If we compute the time constant, it appears paradoxically that we have increased it. But the computed time constant would be that which is required to reach a steady state value limited only by R2 plus the resistance of the selector magnet. This current is about 500 ma and what concerns us is only the first 24 percent of this value. In actual practice the rise time is quite fast. The actual measurement is 2 milliseconds to reach 90 percent of the final value. This is faster than the computed values for the recommended 120v, 60 ma condition, which are 63 percent in 1 millisecond and 8 percent in 2 milliseconds. The measured rise time for an excellent tube terminal unit is 87 percent in 2 milliseconds—also better than that of the suggested circuit. The resulting current rise curve is given in Fig. 4-30B.

This then is the performance we are looking for—a circuit using transistors, operating at a reasonable supply voltage, and giving performance equal or better than the manufacturer's suggested circuit, and the best tube units. This circuit, using two transistors, requires less total power than that needed to light the filament of a 6L6. The final result is incorporated in a terminal unit a small fraction of the size of the original rack mounted monster.

A word of caution should be added on the usual practice of patching in an extra machine by the simple expediency of putting the second printer in series with the first. It is convenient and usually only requires an adjustment of the series resistance to bring the magnet current up to normal value. However, a moment's thought, and it is obvious that this doubles the inductance of the circuit—and doubles the time constant. The waveform only confirms the sad news; it takes twice as long to reach a given current level, and performance suffers. This can be avoided by placing the printers in parallel instead of series; in this case the time constant remains the same and the performance is not degraded. This is not done with tube units because the tubes are normally working at nearly maximum current. With transistors current is no problem and such operation is easily accomplished. In the circuit in Fig. 3-10 the only change necessary is the value of R2. If separate magnet current control is desired, a duplicate of the circuit with a common Q2 switching transistor would do the job nicely without degrading performance.

Nothing has been mentioned thus far about fall time. In actual practice this does not constitute a problem. In prac-

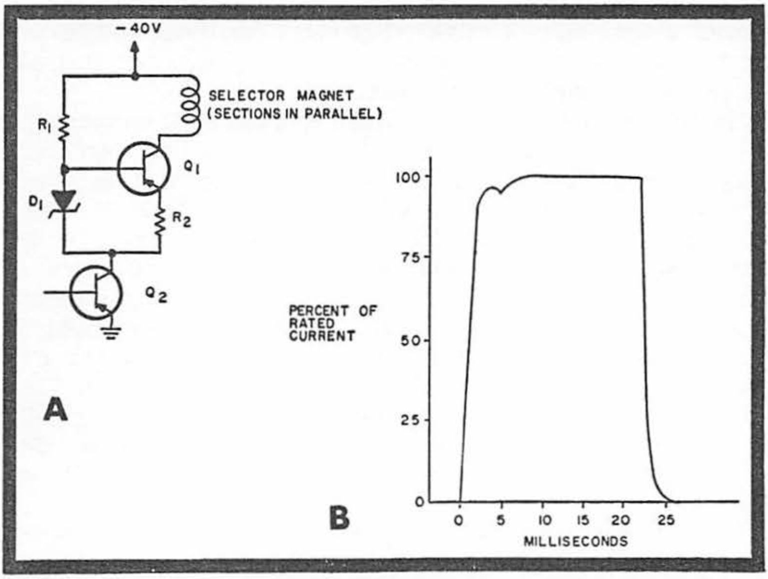


Fig. 4-30. Circuit and steep-sided satisfactory curve using transistor, zener diode, and two resistors.

tically all units, both tube and transistor, the current is down to an insignificant level in 2 milliseconds or less. In no case are voltage spikes observed on any of the waveforms observed. One case where a slow fall is observed is during the testing of various transistors in the circuit of Fig. 4-29A. Transistors ranging from 300 mw up to high power 15 ampere varieties have been tested and almost all performed nicely. Even high power units such as 2N174's and 2N277's worked well. The only failure is an archaic 25 ampere unit which takes about 6 milliseconds to fall to the 10 percent level.

With regard to this process of lengthening the drop out time to compensate for slow rise time, we come across an interesting case. As for practical values, if you want to give it a try R1 is 5K or 10K. R2 in practice is either trimmed to the precise value to give the desired current and left alone—or if like most of us you have an insatiable desire for knobs, use a 50-ohm potentiometer in series with a fixed 22-ohm limiting resistor. The diode should be a low voltage zener on the order of 3 or 4 volts. Those used have been 1N703 or 1N466. Theoretically, the lower the voltage the better, but 6 and 7 volt units have been tried and performed nicely. For transistors, try a 2N251, 2N538 or any 40- or 50-volt unit, the higher the beta, the better.

Frequency Shift Keying (FSK)

CHAPTER 5

One of the simplest problems that faces the new RTTY'er is adapting his transmitter to FSK. Strangely enough, this job seems to be a stumbling block for many. This chapter will cover the principles of one of the simplest and most widely used methods for frequency shift keying. Also, methods for obtaining local copy while transmitting will be discussed.

BASIC PRINCIPLES

In order to transmit RTTY signals, the transmitter frequency is shifted between two different frequencies. The standard method for amateur use is to use the higher frequency for mark and the lower frequency for space. Remember: LSMFT—Low Space Means Fine Teletyping. The standard shift for amateur and MARS use is 850 Hz: the FCC requires the shift to be less than 900 Hz. Many amateurs are experimenting with narrow shifts as low as 100 Hz.

The base idea in shifting a transmitter is to cause the keyboard to switch a reactance in and out of the oscillator circuit in such a manner as to change its frequency. While this reactance can be either an inductance or a capacitance, capacitors are usually used since they are cheaper and have lower loss. To obtain the space, or lower frequency, the capacity is switched across the tuned circuit in a vfo or the crystal in a crystal oscillator. For the mark signal, the capacitor is switched out of the circuit. It would be very difficult to do this switching mechanically, so some type of electronic switching is generally used. A diode makes a very simple and effective electronic switch. By reverse biasing the diode, it looks like an open circuit and by applying a large forward bias, it looks like a short circuit. If the forward bias current is made small, the diode will look like a resistor instead of a short circuit. Thus, by varying the bias, we have an electronically controlled resistance. The switching from reverse to forward bias can be done remotely by the teletypewriter keyboard.

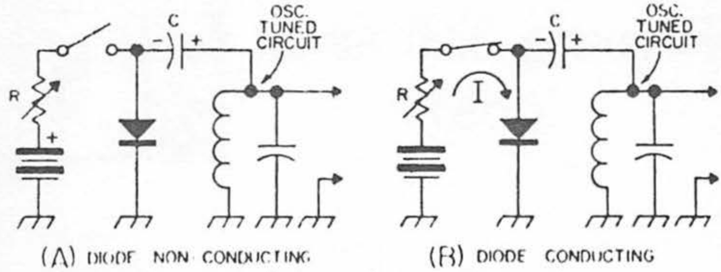


Fig. 5-1. Simplified diode FSK circuit.

Fig. 5-1 shows a very simplified version of a diode FSK circuit which will illustrate how this can be done. The high frequency condition, is shown at A, and the low frequency condition at B.

When the key is open, as in (A), the RF voltage present at the oscillator tank circuit is rectified by the diode, causing the capacitor C to charge up to the polarity shown. This negative bias on the diode causes it to be reversed biased and to look like an open circuit. Thus, the capacitor C is effectively disconnected from the tank circuit and the frequency is determined by the oscillator tuning capacitor. In (B), the key is closed and the diode conducts due to the external battery. The resistor R controls the current I through the diode and consequently controls the effective resistance of the diode. The diode is a nonlinear device; that is, the current through the diode does not change in proportion to the voltage across it. Due to this characteristic, effective resistance of the diode (the ratio of the incremental voltage across the diode to the incremental current through the diode) will change as the DC current through the diode is varied. If this DC current is large, the diode resistance will be very small and the full capacity C is connected across the oscillator tank, lowering its frequency a maximum amount. If the DC current is reduced by increasing R, the diode resistance is increased. A smaller amount of capacitance is effectively across the tuned circuit, and the frequency shift is less.

You may notice one difficulty with the circuit shown. If the key represents the keyboard, then the low frequency would be for the keyboard contacts closed (mark condition). However, we want the space to be the low frequency. Also, when the key

is open, the leads from the diode to the keyboard are floating and might cause trouble with hum pickup. To solve these problems, a circuit shown in simplified form in Fig. 5-2 is used. In this circuit, the key, when closed, cuts off the diode current by bypassing it to ground. A second diode in series will become reverse biased due to a charge built up on C2 which completely isolates the external keying circuit from the oscillator. If shielded leads are used from the diodes to the external keyboard, etc., little trouble with hum or noise pickup should be encountered. When the key is open, forward DC current can flow through both diodes as before, producing the desired low space frequency.

PRACTICAL FSK CIRCUITS

The FSK circuits shown above are quite simple, but there are a few practical problems which need to be considered. The first problem involves the choice of diodes. Vacuum tube diodes are very well suited and are stable and reliable. The 6AL5 and 12AL5 miniature types are commonly used, although the 6H6 and 12H6 are often used in surplus vfo's such as the ARC-5 series. Of course, heater power must be supplied to these diodes. Many of the point-contact germanium diodes work very well if simple precautions are taken. One of the best types is the 1N100 (or 1N99) which has a very high back resistance and is very small. However, the more common 1N69, 1N34A, and similar types will do a very good job. Junction diodes, such as silicon power rectifiers are not too satisfactory, due to their high junction capacitance when reverse biased.

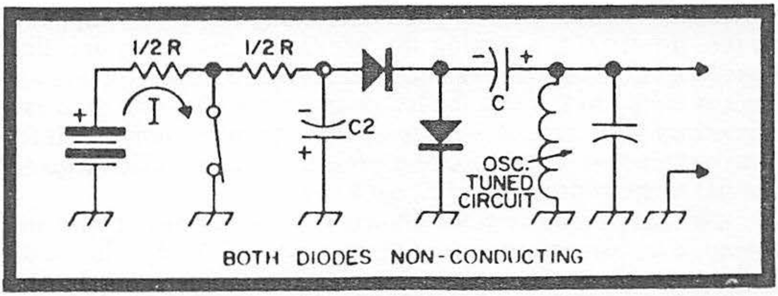


Fig. 5-2. Simplified diode FSK circuit for space-low condition.

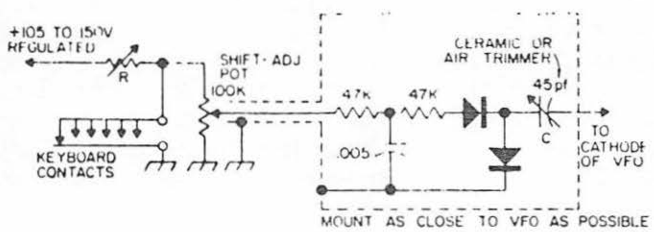


Fig. 5-3. Simple practical FSK circuit.

The second problem is that of eliminating key clicks which can produce excessive interference just as in CW. The problem is easily solved by using a simple RC network to soften or slow the transition from mark to space.

A simple circuit which fulfills these requirements and is easily built is shown in Fig. 5-3. It can be added to almost any vfo and provides smooth, stable frequency shift. In this circuit, the normally-closed keyboard removes the voltage from the diodes and the capacitance C is effectively disconnected from the oscillator. Note that this capacitor, which produces the desired frequency shift, is connected to the vfo cathode, since this is a relatively low impedance point. Most modern oscillator circuits have their cathodes above ground for RF. If this is not the case in your vfo, you can tap the vfo coil a few turns from the ground end. When the keyboard is opened, the diodes conduct partially due to the forward diode current which flows. This causes a portion of the capacity C to appear in the vfo circuit, lowering the oscillator frequency for the space signal. The 47K resistor and .005 mfd capacitor act as the key click filter, since the RC charge time causes a gradual frequency shift (about 1 msec) rather than an abrupt shift. Similarly, when the keyboard closes, the gradual discharge of the RC circuit softens the shift back to mark.

The part of the circuit shown in dotted lines should be mounted as close to the vfo tube as possible. Many vfo's will have room to mount a tie-point strip with the components shown right in their shield cans. To avoid a permanent modification of a vfo, the circuit can be built in a small shield

box mounted near the vfo and the connection to the tube cathode made by wrapping a small piece of solid hook-up wire around the cathode pin. The shift adjustment pot may be mounted externally from the transmitter if desired. It can be on the RTTY converter panel or near the keyboard. A shielded lead to the pot is recommended to prevent noise and hum pickup.

The value of the dropping resistor R will depend on the value of the regulated voltage you have available. This circuit draws only a milliampere or so from this voltage source so this can usually be obtained from a VR tube already in the transmitter or the RTTY converter, or one can be added to any convenient power supply. The following initial adjustment procedure is suggested. Choose R to provide about 50 volts at the top of the shift pot with the keyboard open. Now with the pot wiper at the top, adjust trimmer (shift capacitance) C for slightly more than 850-Hz shift on the lowest frequency band to be used. If insufficient shift is obtained with maximum C, either decrease R to get more diode conduction or parallel C with a small mica capacitor. When proper shift is obtained on the lowest frequency band, the shift can be reduced by use of the shift adjust pot for higher frequency bands where the oscillator frequency is multiplied. To illustrate the effect of varying the diode current with the pot. Fig. 5-4 shows the

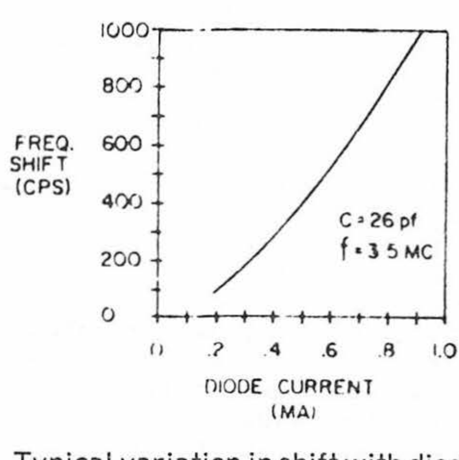


Fig. 5-4. Typical variation in shift with diode circuit.

frequency shift vs diode current for a vfo using this circuit with 1N69 diodes at 3600 kHz.

SHIFTING CRYSTAL OSCILLATORS

Many crystals can be successfully shifted 850 Hz using diode shifters. However, some will quit oscillating before sufficient shift can be obtained. The circuit shown for vfo's can be used with slight changes. The shift capacitor C is connected to the oscillator grid and the shift pot is eliminated. Sufficient current is bled through the diodes to cause them to conduct completely instead of partially. Thus the diodes are acting as switches instead of variable resistors. The shift capacitor C is adjusted to obtain proper shift. Most existing crystal oscillators will be found to be unsatisfactory for shifting due to high fixed capacity across the crystal circuit. It is better to build a new oscillator taking great care to keep stray capacity down, such as using very short leads in grid circuit, etc. A high gain tube such as a 6AK5 is the best choice. The disadvantage of the crystal-FSK circuit is that usually there is no margin for zeroing in on a net frequency since all possible pulling needs to be utilized for shifting. It is possible to adapt the vo circuit that is popular for mobile SSB rigs to provide a tunable crystal-FSK circuit.

OBTAINING LOCAL COPY

When transmitting FSK with the circuit of Fig. 5-3 the keyboard operates only the FSK circuit and does not print local copy. To monitor what you are transmitting, it is necessary to tune your receiver exactly to your transmitter and to allow the RTTY converter to operate the printer. The receiver gain must be reduced to prevent overload. While this method allows continuous monitoring of the transmitted signals, there are some drawbacks. Beside the obvious difficulties in switching receiver gain and returning in case the other station is not right on your frequency, there is a problem in keeping the keyboard contacts clean. These contacts are subject to collection of an oil film along with dust and dirt. In the FSK circuit, they break only about 1 ma of current. This is not sufficient to keep this film burnt off as is the ease when the 60 ma magnet current is being interrupted. Also, all spark suppression filters must be removed from the keyboard circuit when driving the FSK direct or these will cause bias distortion. Then these filters are not available for local loop operation.

A method which gets rid of these problems is to operate the keyboard and printer in series in a local 60 ma loop along with a polar relay. The relay contacts then repeat the keyboard pulses to the FSK circuit and the printer provides direct local copy. A polar relay is used rather than an ordinary single-coil relay which would cause pulse distortion due to its different pull-in and release currents. The circuit shown in Fig. 5-5 is a simple way of using this method.

A DPDT send-receive switch (or relay operated from transmitter S-R relay) changes the printer magnets from the converter for receiving to the local loop for transmitting. With some converters, the polar relay and keyboard can be permanently connected into the printer loop and the keyer tube circuit can supply local loop current for transmitting by means of a hold switch or relay. In mounting a polar relay, be certain to mount either vertically, or, if horizontally, so that the relay armature moves in a horizontal plane.

The relay contacts are enclosed and free from dust and dirt. Clean frequency shift keying is easy to obtain. Another advantage of this method is that it allows the shift to be easily reversed by means of an SPDT switch. This feature is needed for some transmitters using heterodyne vfo's where the upper beat frequency is used on some bands and the lower beat on other bands.

A MERCURY-WETTED KEYING RELAY

A mercury wetted keying relay may be added to the circuit shown in Fig. 4-2 to permit you to key the transmitter

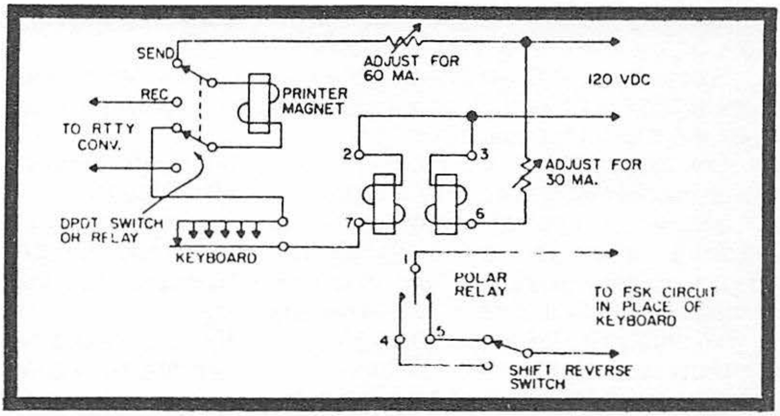


Fig. 5-5. Recommended keying circuit to obtain local loop copy.

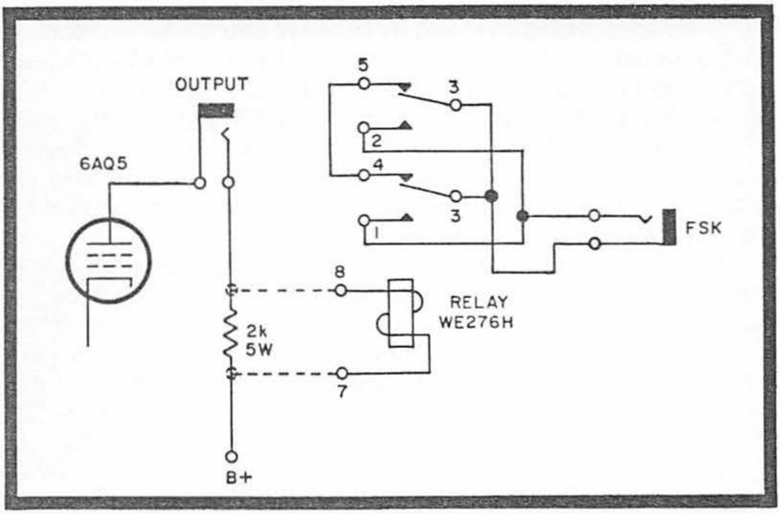


Fig. 5-6. Adding a keying relay to the terminal unit described in Fig. 4-2.

through the keyboard of your machine. Depressing the keys on the machine will open the loop current and permit the relay to key the frequency shift keying circuit of the transmitter and at the same time produce local copy on your machine.

The addition to the circuit is shown to the right of the dotted lines in Fig. 5-6. To the left, the section of the original circuit covering the 6AQ5 is shown.

The mercury wetted relay is the WE276H, although others will work as well, with the required socket changes.

The only word of caution is that the mercury wetted relay must be mounted in a vertical position.

When the modification is complete, let us consider the use of frequency shift keying, and a simple way of accomplishing it on the average transmitter.

Let us see what we mean by frequency shift keying, assuming that we will employ the standard shift of 850 Hz.

The terms applied to this difference in frequency are mark and space, the first being the RF carrier and the difference in this mark and the capacity introduced into the cathode circuit of the bfo, is called the space signal.

To illustrate, take a frequency of 7137 kHz. This would be the mark signal. Now to shift this signal down 850 Hz would produce a space signal of 7136.150 kHz.

Now how do we accomplish the change in our mark signal or the fundamental frequency of our transmitter?

Fig. 5-7 shows a simple circuit which can be used with most transmitters and others with certain modifications, but the principle of creating the frequency shift remains the same.

What occurs is that additional capacity is placed across the LC circuit of the oscillator, which lowers the oscillator frequency sufficiently to move the transmitter carrier frequency 850 Hz.

It should be pointed out that the mark signal is always the higher of the two frequencies, so that the space signal is shifted downward in frequency 850 Hz.

The slug tuned coil (B) is made of 15-20 turns of about no. 22 wire on a ½-inch form and is tuned to an inductance of about 40 mh. If you are unable to reach the 850 Hz between the mark and space signals, vary the slug in (B) slightly.

Adjusting the shift pot must be done slowly, observing the scope monitor for full deflection on both mark and space. A little experimenting with this adjustment will make for an 850-Hz setting.

A VFO FOR FSK

If you operate both teletype and sideband you want a minimum-drift vfo with frequency shift keying provisions. An excellent way to minimize drift caused by heat is to isolate the tuned circuit in one box and the heat-generating components elsewhere (Fig. 5-8). Also, let the vfo run continuously. Keying the oscillator can be a cause of drift.

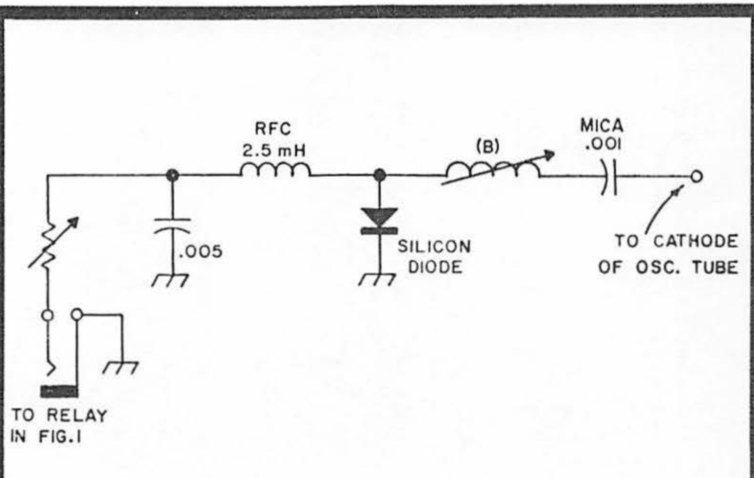


Fig. 5-7 Simple frequency shift keyer adapter for a vfo.

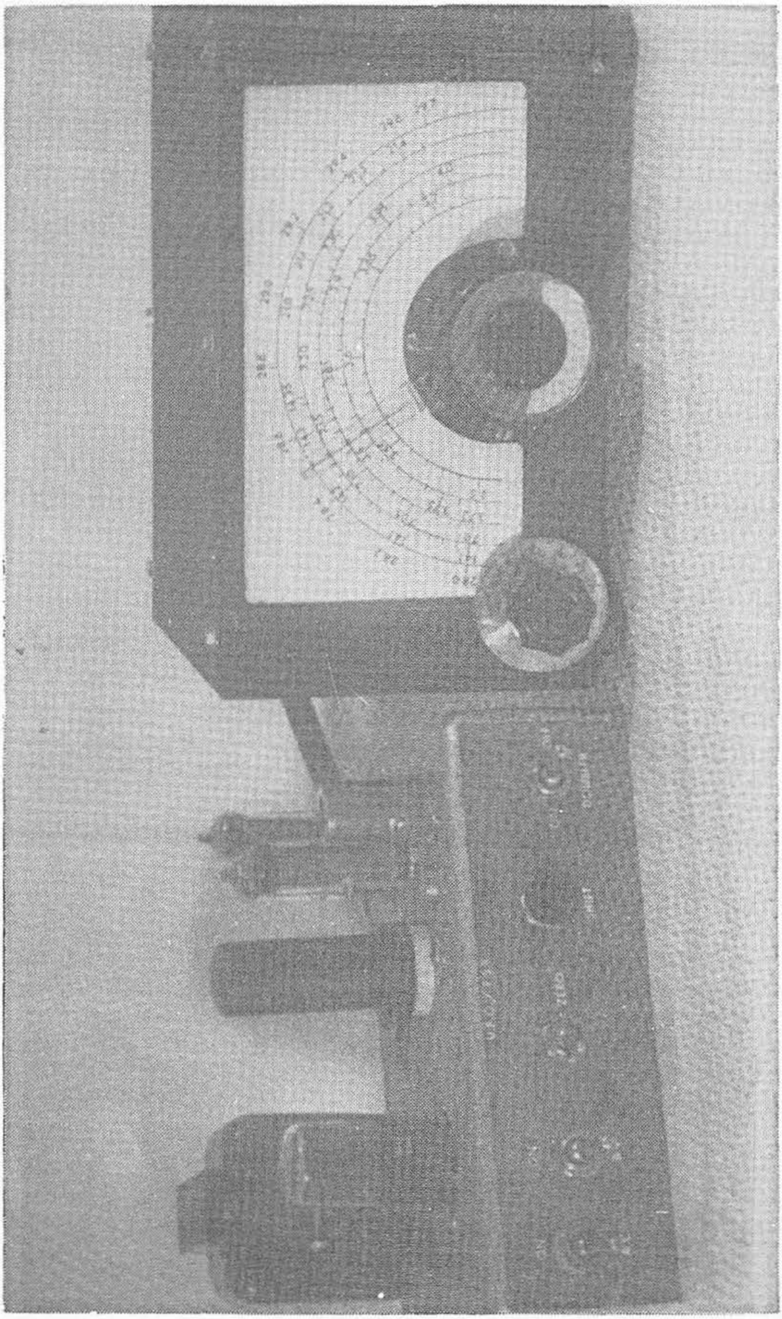


Fig. 5-8. Front view of the two units. The connecting RG-22-U cable is seen in the background.

It is important to use leads as short as possible in the tuned circuit and make then of no. 12 wire, or heavier. If the length of the lead is much over an inch or so, it should be supported at mid-point by a ceramic standoff insulator. Make sure, too, that all components in the tuned circuit box are securely fastened down. You should be able to pound on the surface on which the tuned circuit is placed without having the frequency change.

This vfo uses a series-tuned Colpitts circuit (Fig. 5-9), with a 6AG7 as the oscillator tube. The plate voltage is regulated at 216 volts, and the screen-grid voltage, regulated at 108 volts, is taken from the point between the two OB2 regulator tubes. A little more output can be obtained by increasing the screen voltage to equal the plate voltage, but since this vfo is not intended as a power-generating device, we used the lower voltage for the screen grid. Adequate drive to our exciter, which uses a 5763 buffer and a 5763 buffer-multiplier driving a pair of 6146's, is obtained from 80 through 10 meters.

The coil L2 is slug-tuned to resonate in the 40-meter band. Additional output is obtained on 20, 15, and 10 meters by switching the coil into the circuit. It is switched out of the circuit during operation on 80 and 40 meters, resulting in an untuned output circuit. With most transmitters, the coil should not need to be in the circuit in 40-meter operation.

In this vfo, you allow the vfo to run continuously, you eliminate keying the oscillator. The relay, K1, is used to add 50 pf capacitance to the 6AG7 grid circuit during receive periods. This capacitance takes the signal from the vfo far enough down the band, even on 80 meters, so that you don't run into your own signal when tuning around, unless you are listening 15 kHz (on 80) away from your transmitting frequency.

To zero-beat a station, the DPDT switch, S3, applies 115 volts AC to the relay coil, closes it, and removes the added capacitance from the grid circuit. During transmit periods, the relay obtains its 115 volt AC from the transmitter transmit-receive switch. If your equipment does not contain such a feature, it should be easy to add. Use a DPDT switch to provide 115 volts AC for the vfo frequency-change relay and for another relay, located in the transmitter, which grounds the oscillator and final cathodes during transmitting.

The frequency shift keying circuit is straight-forward. Briefly, the 6AL5 acts as an electronic switch. When the tube conducts, it connects the 33 pf silver-mica capacitor to ground, through the tube. Voltage for the 6AL5 is obtained through the 1-megohm potentiometer, which determines the amount of the shift. The OA2 voltage regulator prevents changes in line

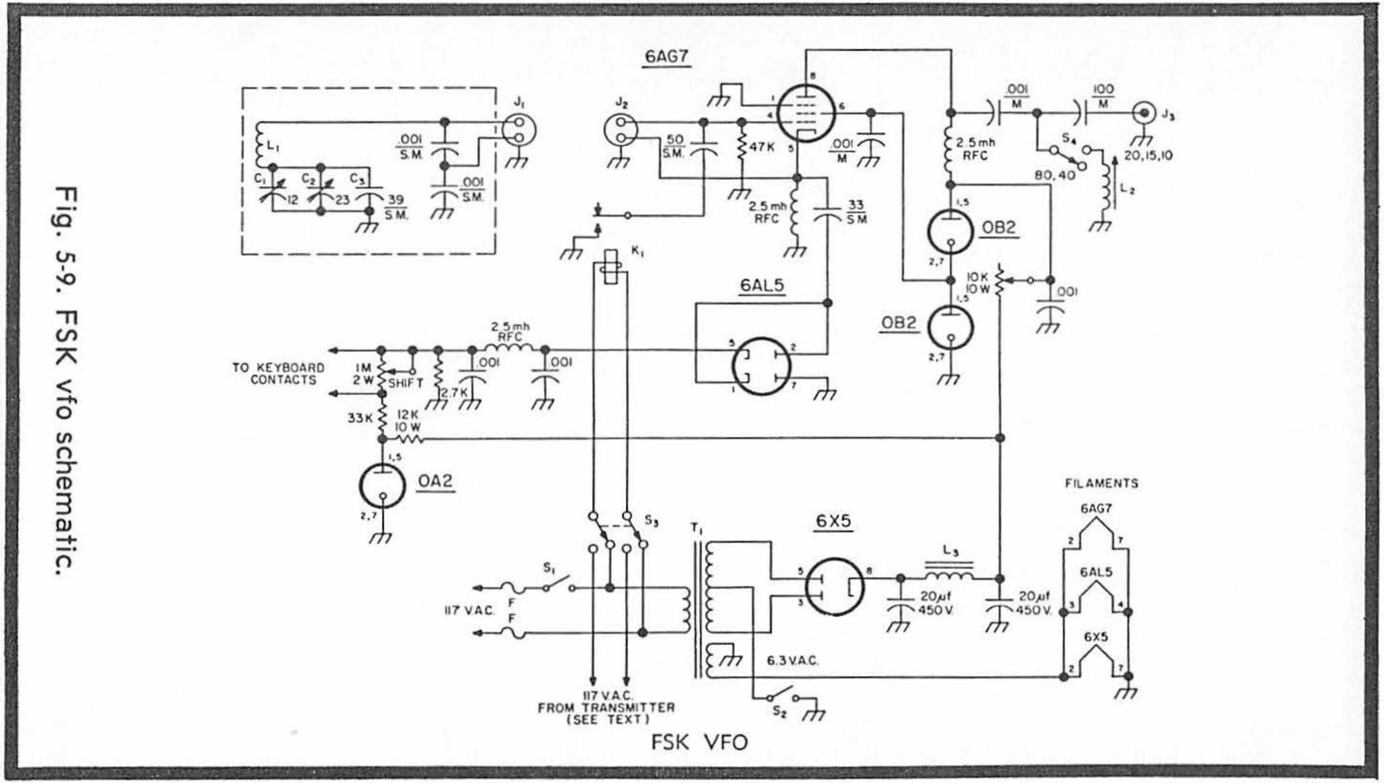


Fig. 5-9. FSK vfo schematic.

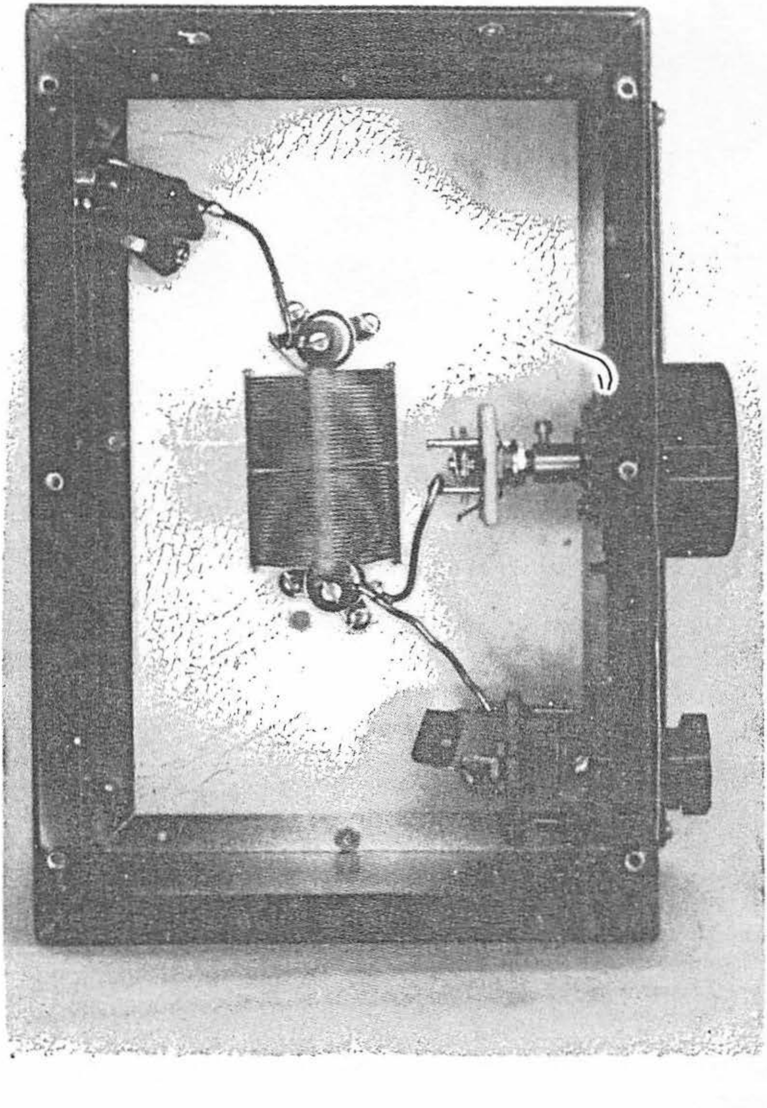


Fig. 5-10. The tuned circuit box. The two .001 mfd silver mica capacitors are mounted at the twin-axial connector.

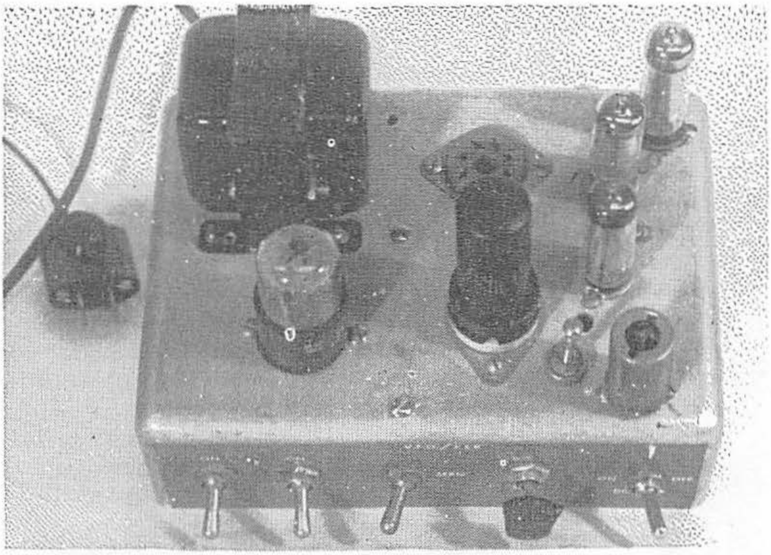


Fig. 5-11. Top view of the oscillator-FSK-power supply chassis. The OA2 voltage regulator tube, for the FSK circuit, is at the right rear of the chassis. The two OB2's, for the oscillator circuit, are to the left and ahead of the OA2. The shield at the front right corner covers the 6AL5 FSK tube. The 6AG7 is located directly in front of the octal socket, which provides connections to the teletype machine. The 6AX5 rectifier is to the left of the 6AG7. The slug-tuned output coil is located to the left of the 6AL5. The AC power connector, in the background to the left, is an Elmenco fused plug.

voltage from affecting the amount of frequency shift once it has been set by the potentiometer.

Using the values shown in the FSK circuit, sufficient frequency shift is obtained on 80 meters to allow 850-Hz RTTY frequency shift keying. The shift potentiometer allows the shift to be varied appropriately for the band in use.

Tuned Circuit Unit

The tuned circuit is built in an aluminum utility box (Fig. 5-10), black hammertone finished, 5" x 6" x 9" (California Chassis No. CAB-5). The coil, B&W JEL-80, has the base and the link carefully removed. The remaining polystyrene mounting rod is mounted on two National GS-1 pillar insulators, which, in turn, are mounted on the bottom of the box. The bandset capacitor, C2, is mounted in the lower left-hand corner of the panel. You will have to drill a hole through the

edge of the National ACN-1 to mount the capacitor. The 39 pf silver-mica capacitor is mounted across the bandset capacitor terminals.

Any good-quality three-terminal connectors can be used on the tuned circuit and oscillator chassis. UG-103 connectors may be used, but these are not polarized. It is essential that the connections between the tuned circuit and oscillator chassis be made as shown in the schematic or the unit will not oscillate. Use small daubs of red paint on the panel and cable connectors to indicate polarization. RG-22-U two-conductor shielded is the cable used.

Oscillator, FSK, and Power Supply Unit

The oscillator, frequency-shift keyer, and power supply (Fig. 5-11) are built on a deep-drawn aluminum chassis measuring approximately 8" x 5½" x 2½".

When building a vfo don't try to save money on capacitors. Buy good-quality silver-mica capacitors for both the vfo and FSK circuits. The familiar, inexpensive ceramic capacitors may be fine in some applications, but they do not belong in any frequency-determining circuit.

The vfo should be calibrated by using a frequency meter. The calibrated dial can be drawn, using a compass and india ink, and the calibration points marked. The numbers are

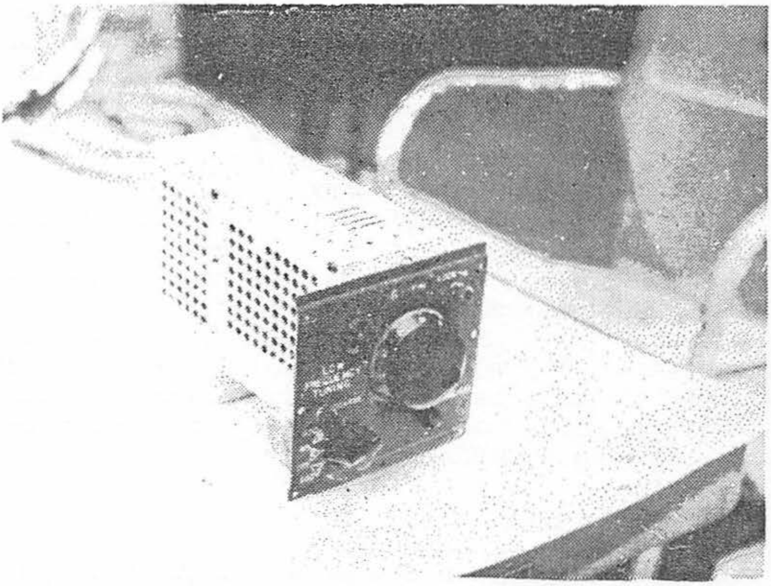


Fig. 5-12. The 0-17-ART-13 pto.

Instant Lettering. The letters and numbers are available in large sheets and are applied simply by burnishing them onto the surface to which you want to apply them.

Parts List

C1—Cardwell PL-6001 with last rotor plate removed.
C2—Hammerarlund HF-35 with last stator plate removed.
J1, J2—UG-103-U panel-mounted connector.
J3—RCA phone plug.
K1—SPST relay, 115 vac.
L1—Barker & Williamson 80-BCL coil with link and base removed.
L2—30 turns no. 26 enameled copper wire wound on ⅜-inch ceramic form, slug tuned.
L3—Broadcast receiver replacement-type choke.
P1—Elmenco fused line plug.
S1, S2, S4—SPST toggle switch.
S3—DPDT toggle switch.
T1—Stancor no PC-8418 power transformer (230-0-230 volts at 50 ma; 6.3v at 2.5a).

FSK EXCITER

The basic concept of this device was borrowed from the Northern FSK keyer, which uses a 200-kHz oscillator beating against a crystal in a balanced modulator. One surplus item available which seems ideally suited to this project is the sub-assembly O-17 ART-13A. This is a low frequency pto (Fig. 5-12) covering the range of 200 kHz to 600 kHz in three bands and is used in the AN-ART-13. The unit uses a 1625 tube, has a very fine, slow tuning dial, and is extremely stable.

(Note: If this pto cannot be obtained, don't throw the idea out. A 400- to 600-kHz vfo is not difficult to construct around a standard 455-kHz bfo coil, and stability is relatively easy to obtain at this frequency.)

Since the pto is the variable portion of the frequency control it is desirable to use it without modification. This could be done by shifting the crystal. It would minimize the unwanted frequencies if the exciter output were one-half the transmitter output frequency.

Construction Stages

Actual construction is straightforward and nothing is critical. Most parts may be freely substituted within a

reasonably wide tolerance (Fig. 5-13). The work and testing may be divided into four stages.

Power Supply

1. The power supply may be any 200- to 400-volt source, 12 volts is necessary for the heater of the 1625. The other tubes may be either 6- or 12-volt types, depending on the supply. A VR150 and a VR105 or equivalent regulator tubes should be used with the required dropping resistor (value depending on voltage). The O-17-ART-13 pto should not be used with more than 105 volts to minimize harmonic output. Once the power supply is assembled, the pto should be connected and checked by listening for its signal on the low end of the BC band.

Crystal Oscillator

2. Next the crystal oscillator should be completed. The only critical parts are the feedback capacitors from cathode to ground and from cathode to grid. The operation of the shift control may be checked by shorting across capacitor C1 with a screwdriver.

3. The two-triode balanced modulator requires no push-pull input of any kind. The tuned circuit L1-C2 tunes the output range desired. Two separate condensers can be used for the sections of C2 if frequent frequency changes are not anticipated.

4. A polar relay is included in the keying circuit. With 30-ma bias current supplied from the exciter power supply, all that is necessary for operating is to plug into the local loop circuit (Fig. 5-14). A turn over switch is provided.

The function switch allows complete station control with one knob. In standby position the cathode of the crystal oscillator is lifted from the ground. In the take-over position, the exciter is switched on by SW2-A, while SW2-B shorts out the loop across the TU output. This prevents the receiver from keying the transmitter (if printer and keyboard are in series), and also allows a quick return of the printer to lower case by flipping the take over switch and punching the letters key. In the third position the C section of SW2 is used to control a transmit relay. In position No. 4, the short across the converter output is removed and the transmitter may be keyed by an incoming signal.

Tuning

As with any heterodyne circuit, care must be exercised not to tune up on a harmonic or wrong beat. The unit should be

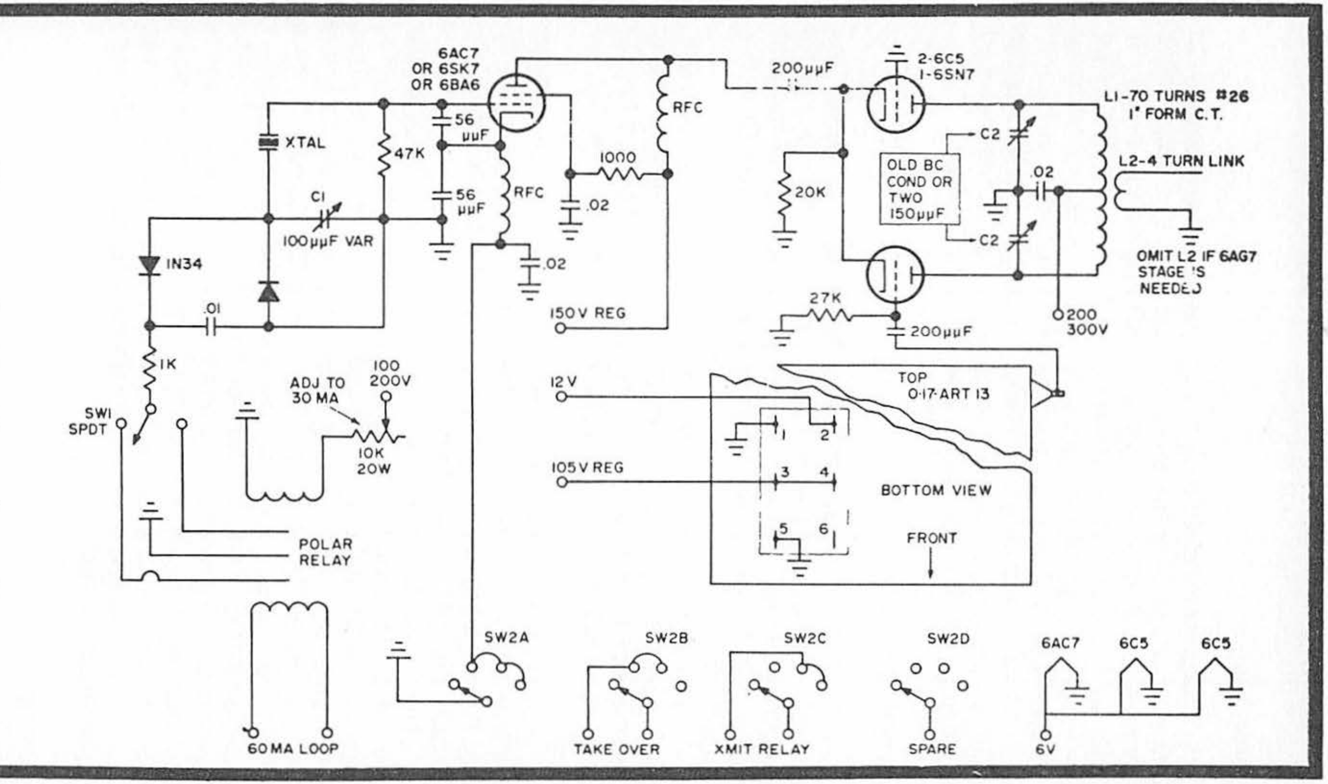


Fig. 5-13. FSK exciter schematic.

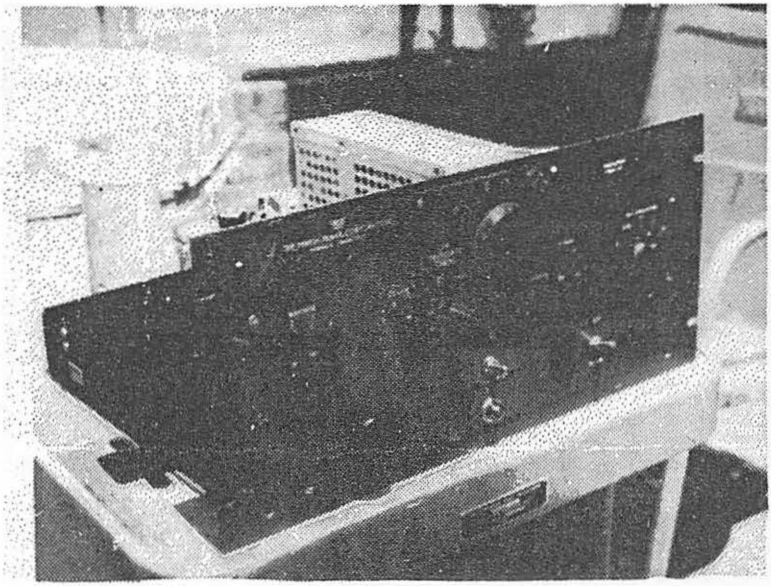


Fig. 5-14. Complete exciter with pto.

set up initially with a grid dip meter or absorption wave meter. The crystal used must be chosen so that the fourth or fifth harmonic of the pto does not fall on or near the wanted frequency.

By doubling in the transmitter any unwanted frequency is further removed from the tuned output of the transmitter.

The average RTTY enthusiast will devote long hours and careful planning to come up with the best possible TU. Then in

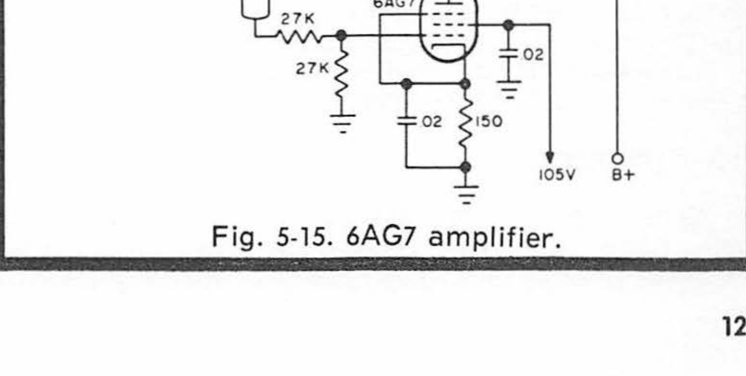


Fig. 5-15. 6AG7 amplifier.

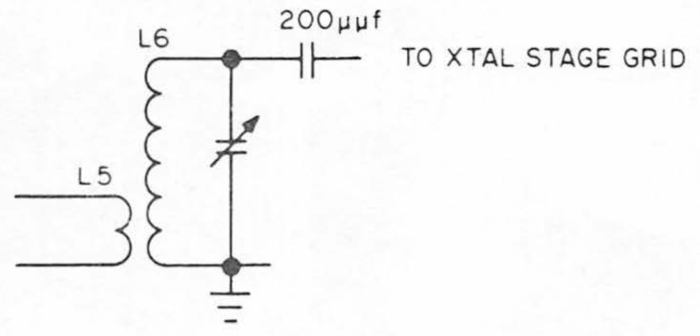


Fig. 5-16. Coupling coil to transmitter.

a rush to get on the air will, all too often, slap a diode shifter on any existing vfo—with less than the desired result. Or, should he decide to shift a crystal he may spend hours grinding the rock to a net frequency only to find his shift shy of the 850 Hz, not to mention that the net frequency may change about the time he finally gets within tolerance.

Some transmitters may require more drive than that obtainable from the balanced modulator. An amplifier stage becomes necessary. A 6AG7 is a logical choice (Fig. 5-15). The grid of this tube is capacitvely coupled to one side of coil L-1 through a short length of RG-58. Two or three turns should be removed from this side of L-1 to maintain balance. (If individual condensers are used for C-1, this may be maintained by tuning.)

The layout of the 6AG7 stage must be made with care. Some physical separation is desirable between the coil L-1, and the 6AG7 tube socket. The output coil, L-2, must be placed above chassis, (if L-1 is below) with the plate lead going directly topside from the tube pin. The 27K resistor insures complete stability.

Low impedance output, either from the balanced modulator, or the 6AG7 stage, if necessary, allows the exciter to be located a convenient distance from the transmitter.

A tuned-link-coupled input coil (Fig. 5-16) should be used at the crystal stage of the transmitter.

FSK WITHOUT RELAYS

All the relays (except selector magnets) in the RTTY system may be eliminated. This includes the formidable polar

relay, a device that is almost as hard to use correctly as it is to obtain.

Since most of the RTTY stations use a vacuum tube keyer for the selector magnets, the only remaining use for the polar relay is to provide the capability of shifting oscillator frequency up or down to achieve a function paralleling that of upper or lower sideband for SSB stations. This function can be achieved with diode keying, thus eliminating all the reasons for using a polar relay. Even the cost of the three diodes required is nominal and the units are available out of most junk boxes and at all electronics parts houses.

Fig. 5-17 is a schematic diagram of the diode keying circuit. The dashed line separating the oscillator tube and the

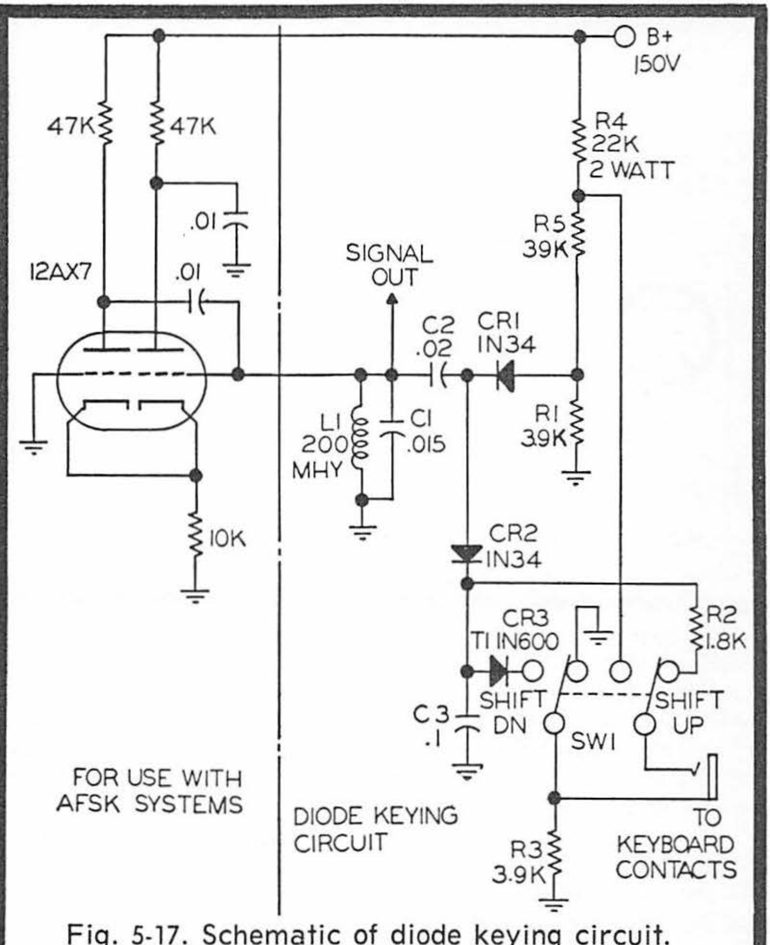


Fig. 5-17. Schematic of diode keying circuit.

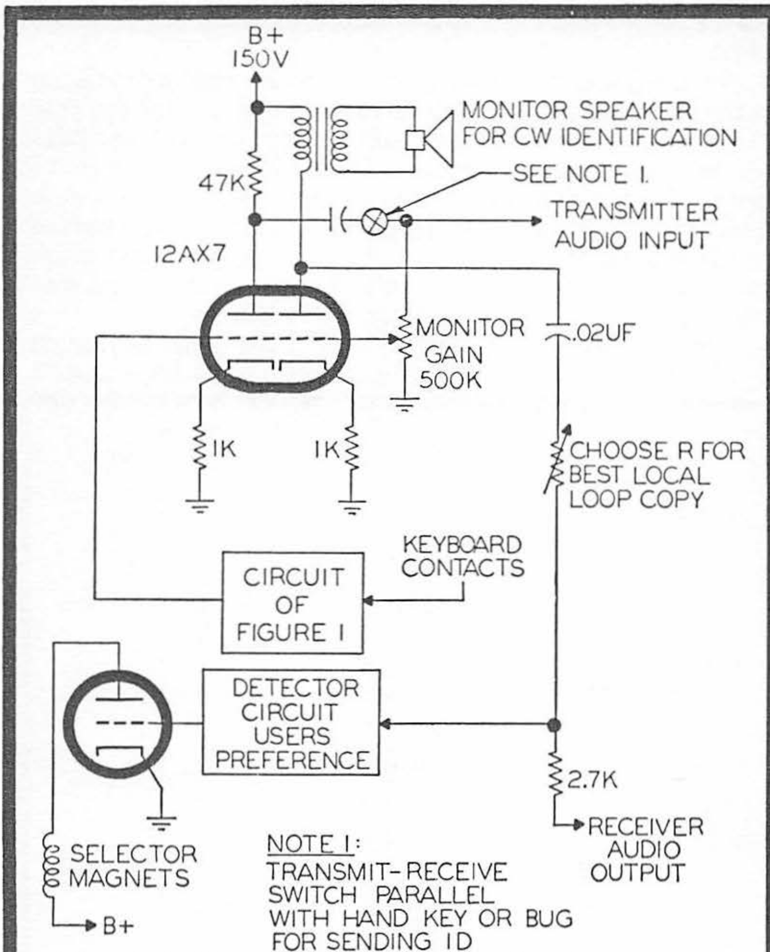


Fig. 5-18. Diagram for obtaining local loop copy.

tank circuit indicates that the circuit may be used for either an AFSK system with the values shown or L1-C1 may be components of the vfo tank circuit for those who prefer keying at the RF frequency. The values shown on the schematic for L1, C1, and C2 are approximate only, and should be selected for 2125 and 2975 cps. For keying the vfo, choose L1, C1, and C2 to provide 850-Hz shift at the frequency desired.

Referring to the schematic diagram, the switch is shown in the more conventional shift-up condition. The keyboard contacts are closed in resting condition, causing a current to flow through diodes CR1 and CR2. CR2 is gated closed and places C2 in parallel with L1-C1 to cause the oscillator to be at

the lower frequency (2125 for AFSK). The .1 mfd capacitor may be considered a short circuit for the frequencies chosen. When the keyboard contacts open, for a mark condition, no current flows and CR2 is gated open. C2 is removed from the tuned circuit and the frequency is determined by L1-C1 only as the higher frequency (2975 for AFSK).

With the switch positioned to the shift-down position, and since the keyboard contacts are closed for space, a positive voltage will be applied to CR3 to prevent current flow through the diodes. The circuit now rests at high frequency. A mark condition opens the keyboard contacts and allows current flow through CR2 causing the oscillator to shift to low frequency.

When used with a 150-volt power supply, current flow through the diodes is limited to 2.5 ma by the bleeder resistor networks, R1, R3, R4, and R5, R1, R2, and R3 were chosen to provide proper frequency shift in both switch positions, eliminating circuit interactions. CR3 should be a silicon diode, while CR1 and CR2 should be germanium diodes. Typical types are shown on the schematic but most available units will work satisfactorily.

Fig. 5-18 details the means for obtaining local loop copy. V2 provides input to the detector circuit from the AFSK oscillator and isolates the transmitter input from the receiver audio output. If selector magnets are keyed through a vacuum tube, no relays will be required in the system.

If the vfo is shifted, the receiver will serve as the local loop in the same manner used to monitor CW signals when a monitor is unavailable.

CHAPTER

Audio Frequency Shift Keying (AFSK)

When you put an SSB transmitter on RTTY, it is apparent that using the microphone pickup for AFSK input is the most desirable method. While there are dangers in this method concerning noncompliance with FCC regulations regarding purity of emission, the advantages of being able to use the VOX circuits for switching and being able to transceive on RTTY make this approach worthwhile. If the transmitter uses a steep-skirted bandpass filter, such a system could yield good results. Indeed, the method of CW generation is introduction of an audio tone into the audio stages.

Desirable qualities for an AFSK circuit are:

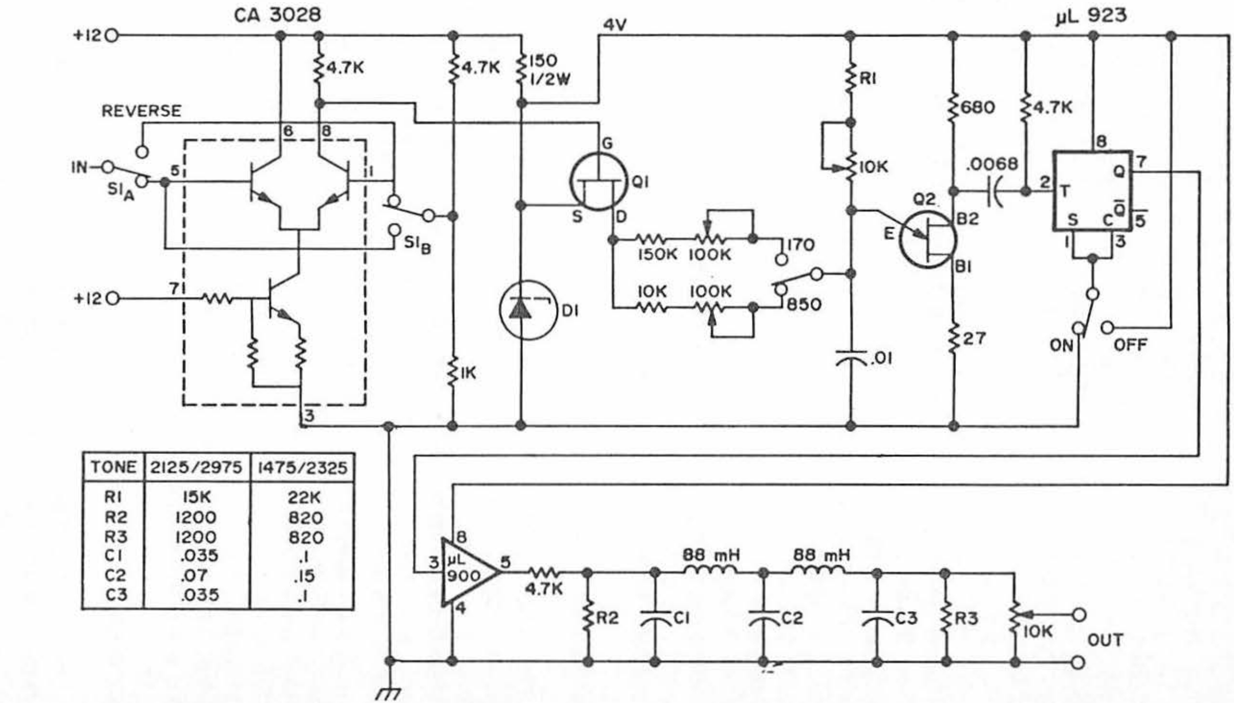
1. Negligible harmonic generation
2. Freedom from keying transients
3. Ability to reverse shift
4. Equal mark and space amplitude
5. Wide and narrow shift capability
6. Ability to use low and high frequency tones
7. Easy to build and align

AFSK WITH SSB

The circuit (Fig. 6-1) is a result of this analysis. Construction may be simplified by the use of a printed-circuit board. The use of inexpensive integrated circuits further simplifies construction. Information is given for use with 2125-2975 and 1475-2325 Hz tone pairs. The latter tone pair was chosen as optimum from the standpoint of minimizing spurious signal generation in a transmitter with a 2.1-kHz bandpass filter.

Circuit Description

A free-running unijunction transistor multivibrator is used as a frequency-shifted pulse generator running at twice



TONE	2125/2975	1475/2325
R1	15K	22K
R2	1200	820
R3	1200	820
C1	.035	.1
C2	.07	.15
C3	.035	.1

Fig. 6-1. Circuit diagram for the AFSK unit.

the desired frequency. This pulse train is divided by two in an IC flip-flop, forming a constant-amplitude square wave of the desired frequency. The square wave is then filtered in a five-pole Butterworth low-pass filter which suppresses all odd harmonics above the fundamental frequency (even harmonics are not present in a square wave). The frequency of the oscillator is shifted by switching an additional resistance from supply voltage to the emitter of the unijunction. This provides more charging current to the capacitor and results in a higher-frequency pulse train. The actual switching is done using a P-channel FET as a switch to achieve the very high off resistance necessary for easy adjustment of the shift. The FET is driven by a differential amplifier which senses the keying loop circuit. This diffamp is used as a current-switching discriminator and allows us to reverse shift very easily by interchanging the input leads. The switching threshold is approximately +2v at the input, which may be conveniently obtained from the TTY loop supply current passing through a series resistor.

The power supply (not described here) is merely a single-ended 12v supply. Regulation is not particularly critical, and the current drain is only 55 ma.

Construction

The AFSK unit, with the exception of the power supply, frequency determining potentiometers, and input coupling circuit, is built on a plug-in printed-circuit board. Pay particular attention to align the key on the semiconductor components with the key on the circuit board and you cannot go wrong. If you are not using a printed board, pay close attention to the basing diagrams given in Fig. 6-2. The 88-mh coils are the surplus telephone loading toroids and are readily available at very reasonable prices.

Since reversing shift is not normally done after initial installation, the switch for this function may be omitted if desired. The plug-in socket may be wired so that turning the board over in the socket reverses the shift.

Alignment

After applying power to the unit from a suitable source the operate-standby switch should be positioned to OPERATE. With the reversing switch in normal, the 25K potentiometer should be adjusted for the low tone. Next, apply approximately 4v to the input: a jumper from the cathode of the

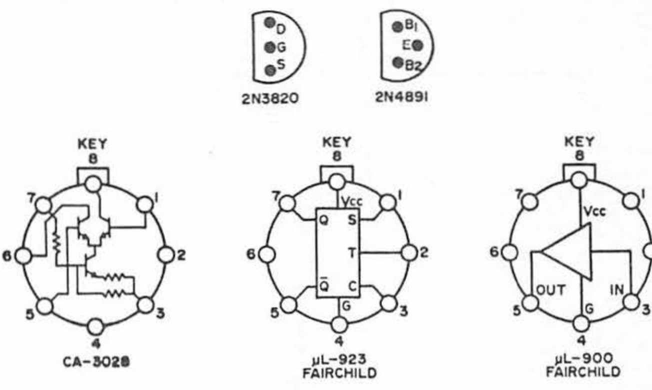


Fig. 6-2. Base diagrams.

zener to the input will do. Now adjust the 100K potentiometers for the narrow and wide shift high tones.

Keying Circuit

The keying circuit is shown in Fig. 6-3. Other schemes can be used; however, they should be examined to make sure that they will not place over 6v on the input of the differential amplifiers..

This AFSK unit represents an advance over many other designs. Its freedom from keying transients and harmonic generation should help clean up some of the signals heard on the band. The presentation of a design for the lower tone pair allows use with many transmitters without modification. Economy semiconductors and integrated circuits are used, and the cost of building the unit with all new parts should be about $10.

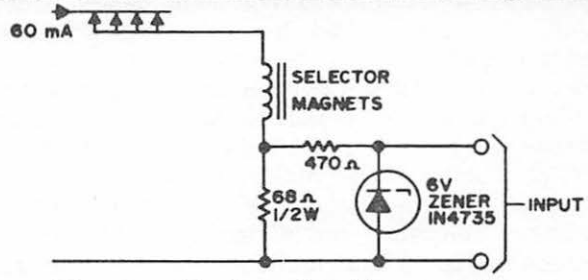


Fig. 6-3. Keying circuit connection.

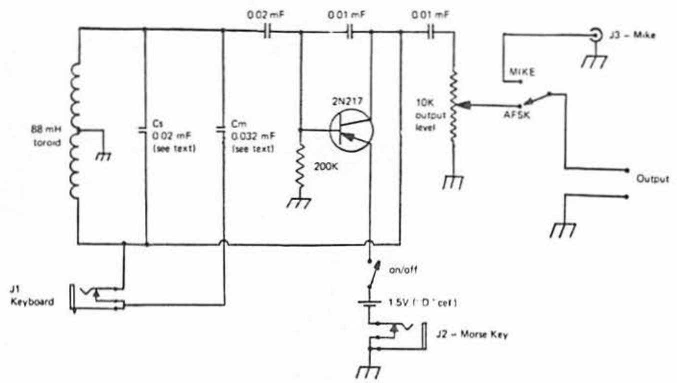


Fig. 6-4. Schematic of AFSK tone generator.

A MAGNETIC REED AFSK TONE GENERATOR

The audio frequency shift keying (AFSK) generator described here uses a bare minimum of parts (Fig. 6-4). It gives a reasonably good sine wave output and, using the keying technique included, a minimum of transients or bias. The keying technique referred to is the application of a new device, the magnetic reed relay, to a Teletype keyer. The reed relay can be used not only to key this AFSK generator, but standard FSK units as well.

Simple circuits published previously have contained at least two transistors and a reasonable amount of other software. The schematic shows this oscillator to have only one transistor and a minimum of other components. The actual audio shift is accomplished by switching an additional capacitance in parallel with the toroid to lower the tone on mark. On space, the switch is open and a higher tone (from less parallel capacitance) is generated. Although Cs and Cm are shown as one capacitor each, they are in reality several capacitors paralleled to give the desired resonant frequency. For such work, you cannot rely solely on the marked value of the capacitor. Tolerance variations can play havoc with combinations set up in that manner. Rather, true trial and error should be used to determine the values after the rest of the circuit is completed.

Frequency Determination

For newcomers, the use of Lissajous Figures is the most practical method of audio frequency determination. Briefly, setting up the circuit shown in Fig. 6-5 will allow frequency matching to within a small percentage. The source of the standard frequency will be investigated later. For the time being, such a standard is merely a known-to-be-accurate signal source. While observing the scope face, try various small capacitors of the approximate suggested values (or use a capacitor decade box if one is available) until a circle or ellipse appears on the screen. When that happens, the two audio sources are within one hertz of each other.

Another method may be used for those lacking an oscilloscope. This technique, almost forgotten in amateur circles, is one using audio beats to match frequency. It permits matching to within one hertz, and gives a better result than Lissajous Figures for tones separated by several hertz. The audio beat method involves feeding the standard tone into a loudspeaker and the AFSK generator to be calibrated into another (both through amplifiers, of course). As the generator frequency is adjusted to that of the standard, using trial capacitors or a decade box—as above—audio beats, that is, variations in amplitude, will be heard. When the beat frequency is less than one or two per second, the two audio tones are matched within one hertz.

Since exact frequency tolerance is not needed, it makes little difference to the receiving operator whether your mark is 2125.0 Hz or 2126.7 Hz. The audio beat method is perhaps,

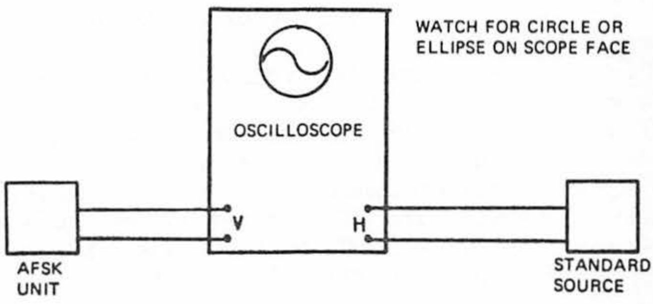


Fig. 6-5. A simple frequency matching setup.

better suited to the newcomer with only a limited audio reference source, at say, five hertz separation between tones, the oscillographic Lissajous Figure will be a meaningless blur, while audio beats will be present at a rapid rate. Thus, adjusting the tones by the beat method, once slow beats are obtained, the frequency is within fair limits.

Frequency Standards

Although much has been said so far about the standard frequency, details have not yet been given. Such a standard should be capable of supplying the mark frequency, 2125 Hz, and the space frequency, 2975 Hz, at a reasonable level of accuracy, stability, and sine waveform. Perhaps the easiest standard to use is another AFSK generator, known to be accurate. Such a generator could be borrowed from a friend or its signal used over a strong VHF link. Alternatively, a calibrated audio generator could be set up for the 2125-Hz mark and 2975-Hz space tones. A third method is available to the amateur who has an operating demodulator. Tune in a strong carrier on the receiver, as from the transmitter's spot function or the 100-kHz crystal calibrator, and adjust the bfo to give the proper tones for Teletype. To calibrate the generator, adjust the standard to produce the tone for space (2975 Hz). Adjust Cs until the AFSK unit produces the correct space frequency, as measured by either of the frequency matching methods above. Close the switch grounding Cm and move the standard to the mark tone (2125 Hz). Adjust the value of Cm for the correct mark tone. Recheck the space tone to be sure interaction has not changed it. Feed the signal of the AFSK generator into the mike input of your transmitter, and adjust the level control for 100 percent modulation, connect a key at J2 or a mike at J3 (for the CW or phone identification required).

If you connect this directly to your keyboard contacts you may have some problems. First of all, the generator must be keyed "dry." That is, there can be no current or voltage on the keying line. Second, the contacts used to key must be clean. While not so much of a problem with a keyboard, the rotating contacts of few TD's are clean enough to key this circuit without hash. The solution to these problems is to key through a set of contacts separate from the Teletype loop.

Magnetic Reed Relay

For years, the only way to do this was by a polar relay. Other, spring-returned, mechanical relays differed too much

to their make vs break current to key without bias (changing the length of the 21 msec Teletype pulses). Recently, however, a new device—the magnetic reed relay—has come upon the scene. Literature has described this device as being able to operate upwards of 2000 Hz. If a 2000 Hz signal were fed to the activating coil, the contacts would make and break 4000 times a second. Each actuation would be 0.25 msec long. It appears, therefore, that the relay could operate on the 21 msec Teletype pulses. Such a relay was procured from a local supply house, and a coil wound on the form enclosed with the relay. The coil has several thousands turns of 32-gage wire, and a DC resistance of 45 ohms. It readily pulls in on 40 ma, thus the 60 ma local loop current assures positive action.

Transmitting tests using the relay to key the low frequency FSK rig showed no detectable bias. The relay was then installed into the loop as shown in Fig. 6-6. The magnetic relay is now used for keying both the AFSK and FSK rigs, with excellent results.

Construction techniques are entirely up to the builder. The AFSK oscillator may be built on a small piece of perf-board and mounted in a minibox with its complement of jacks and switches. The magnetic reed relay, could be attached to a piece of printed circuit board for support, then mounted behind the Teletype patch panel.

The circuit shown here is a fast and easy way to transmit on VHF Teletype. It is not recommended that the output of this particular AFSK generator be fed to a sideband transmitter to attempt low frequency FSK. The output of this circuit, while a reasonably good sine wave, is not perfect, and some spurious sideband generation may result. The relay keying circuit is

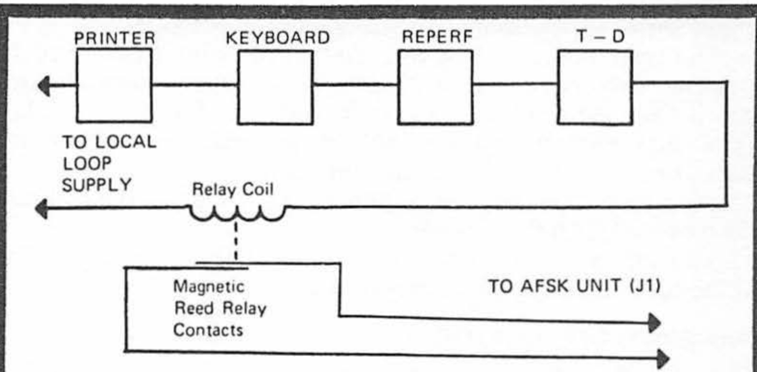


Fig. 6-6. AFSK keying arrangement installed in loop.

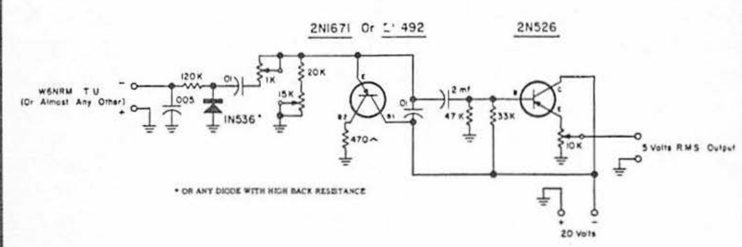


Fig. 6-7. Schematic of a transistorized AFSK.

applicable to all phases of Teletype, and can be used in preference to a standard polar relay. If reversal of the tones or keying mode is desired, SPDT reed relays are available. The total elimination of a bias supply (needed with a polar relay) and the ability to separate receiver and transmitter from the same frequency and still maintain local copy are easily appreciated.

A TRANSISTORIZED AFSK

An AFSK unit using a unijunction transistor develops a sawtooth oscillation, not a sine-wave, but works satisfactorily on 2 meters.

The unit may be built on a breadboard and operates on a self-contained battery supply.

In operation, merely unplug the frequency shift cable running into the transmitter and insert it into the transistor AFSK. Adjust the frequency shift control on the TU to give the proper shift as indicated on the tuning eye and scope.

The unit, shown in Fig. 6-7, uses a 2N492 transistor which is rather expensive, but a 2N1671 Texas Instrument should work as well. The voltage is 20 to 30 volts DC. Caution must be taken with the unijunction not to ground the emitter in operation, for this will damage the junction.

Five volts rms was developed on the audio output, which will drive any audio circuit.

The unit may be used as an MCW oscillator by breaking the battery supply and inserting a key.

TWO-TONE OSCILLATOR

This oscillator has some of the same design characteristics of other oscillators except that transistors are used

rather than vacuum tubes. The oscillator discussed here can be keyed on either mark or space pulses.

AFSK operation requires a reasonably stable audio oscillator that will supply standard tones of 2975 cycles for space and 2125 cycles for mark when keyed from the keyboard of the teleprinter. The two output tones are simply fed into the input of a modulator or into an SSB exciter which has very good unwanted sideband and carrier suppression.

Transistor Q1 (Fig. 6-8) is the oscillator. The oscillator inductor is an 88-mh toroid type telephone loading coil. This inductor is noted on the schematic as L1-A and L1-B. Notice that capacitors C1 through C8 are utilized for tuning the oscillator and should be high-grade mylar, paper or mica. Do not use disc ceramic types. When adjusting the oscillator frequency, be sure the entire circuit is complete and the keyboard is connected in the circuit. Tune the space frequency (2975 cps) with the keyboard circuit open. Tune the mark frequency (2125 cps) with the keyboard circuit closed.

Circuit Description

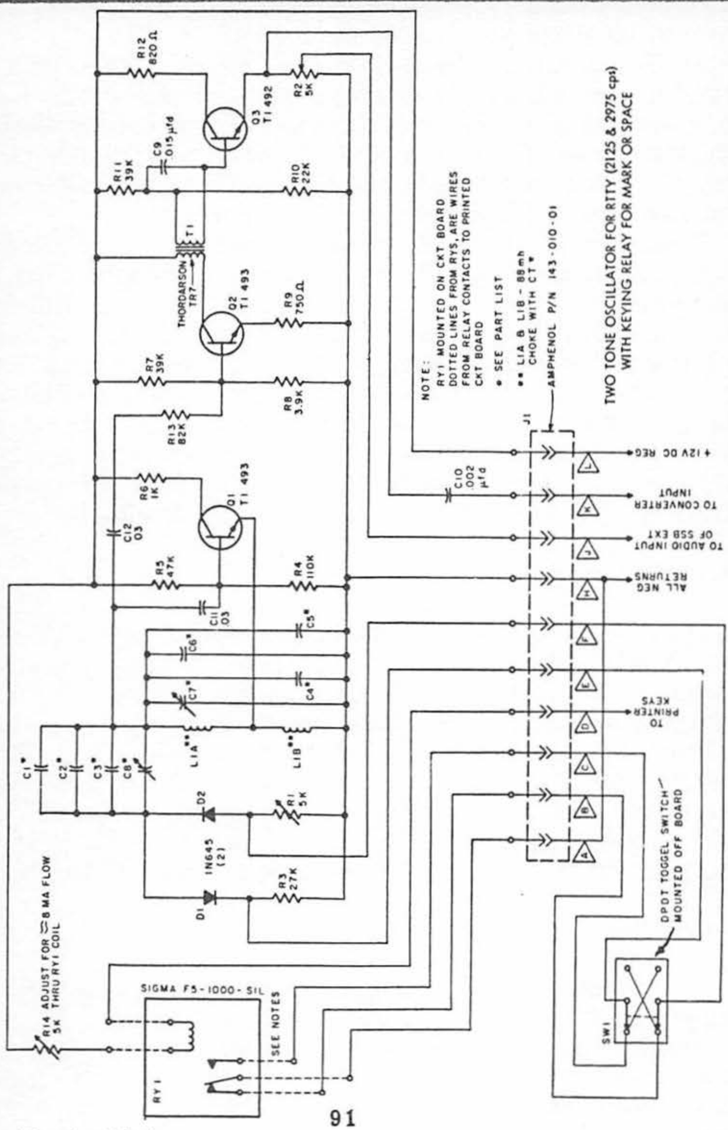
The output from the oscillator, Q1, is RC coupled to the base (input) of Q2, which serves as both an amplifier and isolating stage. The output of Q2 is coupled through an interstage transformer to the base (input) of emitter follower Q3. From the output stage, Q3, the audio tones are fed into the converter, for local copy, and into the modulator or SSB exciter. Converter input is taken directly from the emitter of Q3, while the output level for the transmitter, may be adjusted and is coupled out from the arm of the emitter load resistor, R2.

Construction

The two-tone oscillator is quite simple to construct on a printed circuit board. Only a small drill, small soldering iron and simple hand tools are needed. Provisions may be made to mate the printed circuit board with a connector for plug-in use, or the circuit board may be wired directly into associated equipment. All parts listed for this project may be purchased from your local distributor. However, a well stocked junk box should produce most of the parts required to complete this circuit. The 88-mh coils are readily obtainable from surplus.

Adjustment

The relay is set for a pull-in current of approximately 8 ma. However, other relays with equal coil values may be used



Parts List

All resistors ½ watt

All condensers 100 volts

C1, 2, 3 plus C8 equal approx. .0398 mfd (2125 cps tone)

C4, 5, 6 plus C7 equal approx. .0347 mfd (2975 cps tone)

L1A, L1B—see text

Fig. 6-8. Two-tone oscillator for RTTY.

in this circuit. Relay current is adjusted by potentiometer R14. This adjustment is made by breaking the circuit between R14 and one side of the relay coil and inserting an appropriate meter.

Oscillator output level is adjusted to obtain equal output amplitude for both tones. This is done by keying the printer and adjusting R1 until there is no difference in the transmitter plate current readings during mark and space times. When this adjustment is completed, tone output of the transmitter will be equal for both mark and space.

A NOVEL AFSK OSCILLATOR

The standard capacity-switched LC oscillator leaves much to be desired. Switching a discharged condenser into an oscillating circuit causes a pause in the oscillation, resulting in a shortened mark pulse. Switching transients above the normal output level were seen, which can cause overmodulation or require operation at reduced modulation levels. The switched-C oscillator does not normally have equal mark and space output levels, and cannot easily be adjusted to exact frequency. It does have the advantages of simplicity and excellent stability.

If the added mark capacity were very small compared to the normal circuit capacity, the oscillator circuit Q and impedances would be nearly constant and there would be no switching problem. A large shift can be obtained from a very small added capacity only if the operating frequency is high, as in a beat-frequency audio oscillator. Such a shift bfo produces an excellent switched waveform having easily adjustable shifts. Unfortunately even a carefully designed transistor bfo has very poor long term drift problems that make it unsuitable for RTTY work.

There is one type of oscillator admirably suited to AFSK service, but apparently never used. It has good stability, constant output, shifts instantly and without transients, and is quite easy to shift and adjust—the relaxation oscillator!

Look at the characteristics of two relaxation circuits, Fig. 6-9. The neon bulb is probably the more familiar, but the unijunction is more stable and will be used in this oscillator. The output amplitude is determined by the device characteristics, not by the frequency-determining R and C. The frequency is easily and smoothly shifted if the charging resistance R is changed. The sawtooth output is easily smoothed into a sine wave by a low-pass filter.

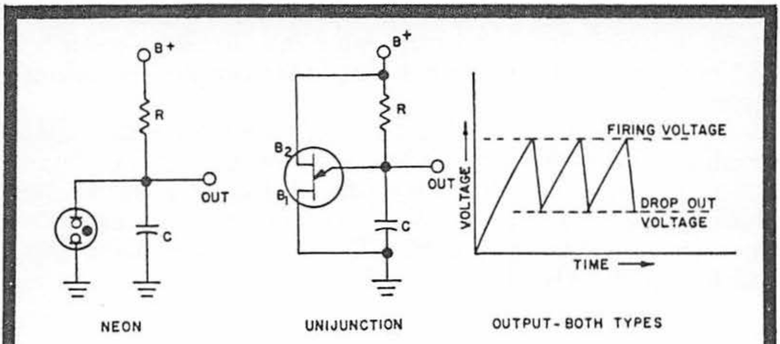


Fig. 6-9. Relaxation oscillators.

Fig. 6-10 shows the circuit of an AFSK relaxation oscillator. The unijunction transistor is an inexpensive and stable solid-state equivalent of the neon bulb. Its operation will be described only briefly here.

Operation

The unijunction is operated with +20 volts on base two. The emitter is an open circuit for voltages below about 10 (varies with the transistor). As condenser C1 charges through resistors R1 through R5, its voltage bulds up to 10 volts. At this voltage, the emitter to base-one junction breaks down, much as the gas in a neon bulb, and discharges C1. This cycle repeats, producing the familiar sawtooth voltage across C1. If the charging resistance is returned to the same voltage point as base two, the relaxation frequency is remarkably independent of voltage and temperature variations.

The generated sawtooth voltage is isolated by emitter-follower Q2 and fed into a low-pass filter. As seen in Fig. 6-11, the filter cuts off above the space frequency (2975 cps) and is almost 40 db down for all harmonics of both the mark and space frequencies. The output waveform is a very good sine. The input resistor, R12, can be set so that the output voltage varies less than ½ db over the range of 2125-2975 cps.

Inductors L1 and L2 are the familiar telephone loading toroids. L3 is a toroid removed from a surplus filter. Condensers C2 through C5 should be low loss units such as mica or polystyreme selected for proper value on a bridge.

After filtering ,the tones are amplified by Q3. R14 is set to prevent overloading of Q3 at maximum level.

Keying is accomplished by saturating (mark) or cutting off (space) transistor Q4. With Q4 cut off, Q1 oscillates at a space frequency determined by the sum of R1 through R5.

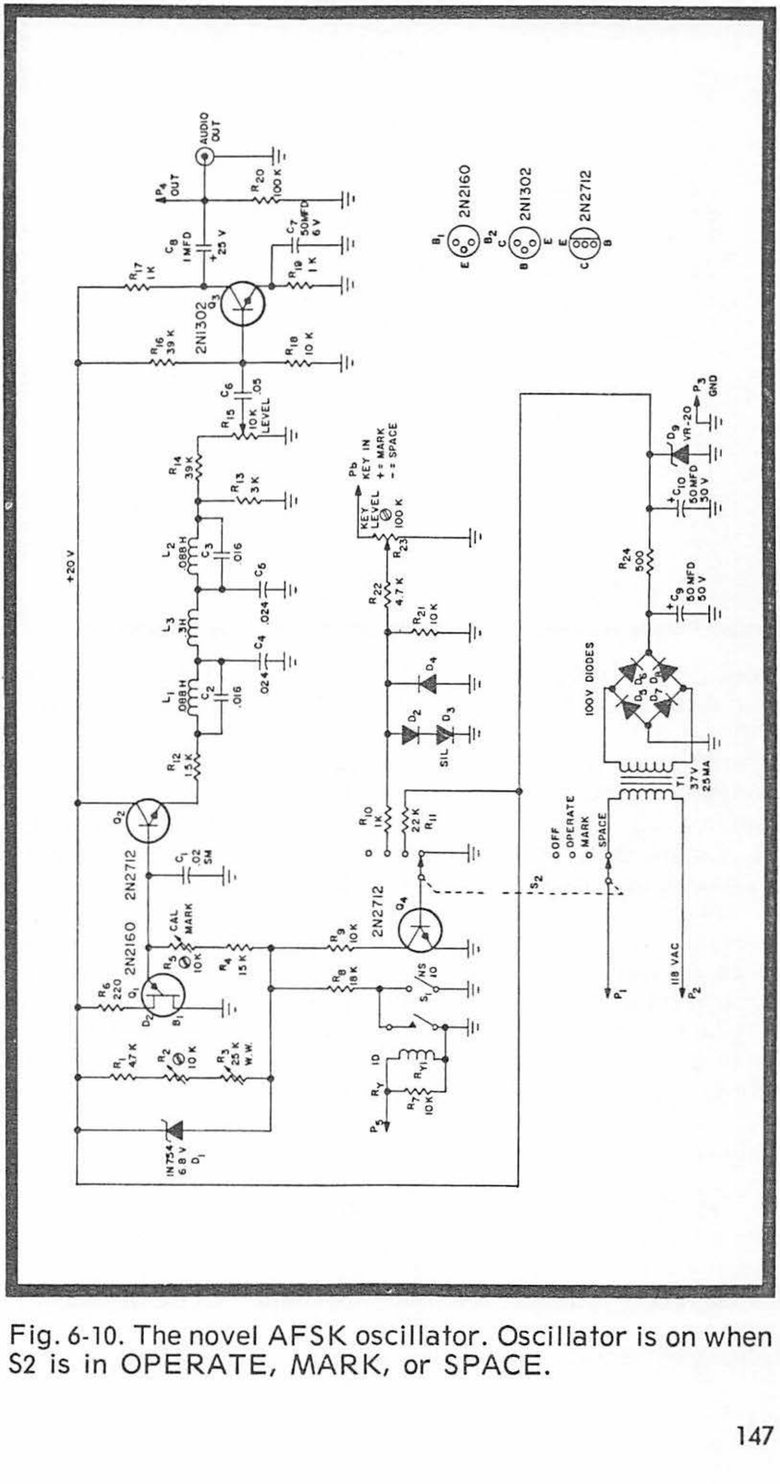


Fig. 6-10. The novel AFSK oscillator. Oscillator is on when S2 is in OPERATE, MARK, or SPACE.

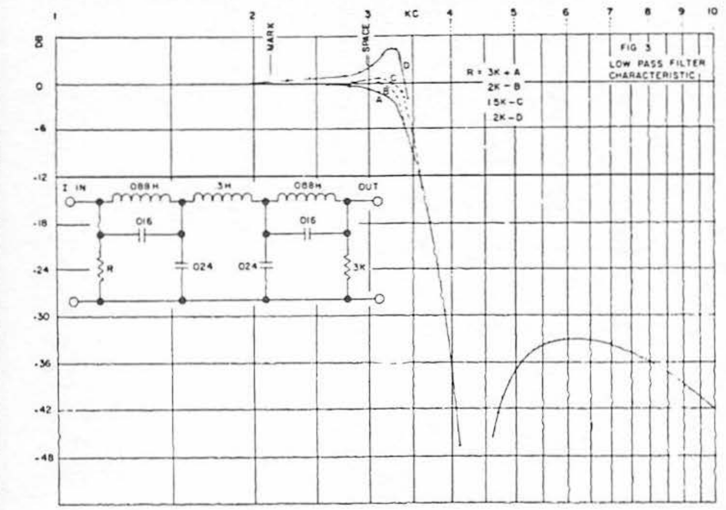


Fig. 6-11. Low-pass filter characteristics.

Diode D1 does not conduct. With Q4 saturated, R8 causes D1 to conduct at a constant 6.8 volts regardless of the setting of the shift pot R3, and Q1 now oscillates at a mark frequently determined only by R4 plus R5. The shift is adjustable from 100 cps minimum to 850 cps maximum by rotating a potentiometer, but the mark frequency is held at 2125 cps by diode D1. Narrow shift for identification is provided by push switch S1 or a miniature relay for remote ID switching.

Q4 is keyed to mark by a one-milliamp positive input current from any source over five volts. The change from space to mark is very sudden and non critical. Diodes D2 through D4 protect Q4 from loop voltage transients. The input may be connected across shunt resistors in the loop (100 ohms for 60 ma loop; 300 ohms for 20 ma loop) or directly to the cold side of a positive voltage keyboard.

Calibration

Set S2 to mark (Fig. 6-12) and adjust R5 for a 2125 cps output. Now set S2 to space, turn R3 for highest frequency and adjust R2 for 2975 cps. Because of variations in unijunction transistors it may be necessary to change C1 to obtain the proper frequency range. With S2 still in space, R3 may be calibrated directly in cycles of shift down to about 100,

depending on the exact characteristics of Q1 and D1. R12 is selected for constant output as the shift pot is varied over its range. With the oscillator connected to the loop, increase R23 just past the point where marking occurs.

A SOLID-STATE RTTY SYSTEM

This unit provides a high performance solid-state AFSK oscillator and tuning unit for RTTY. Silicon controlled rectifiers are used to drive the printer magnet. Several of the earlier amateur-designed AFSK units generate a non-sinusoidal signal and then depend on filters to eliminate all but the fundamental component. The phase-shift oscillator described here has the good points of the earlier units—equal mark and space output levels, no switching transients, isolation from the keyboard, a simple shift system—plus the advantages of lower cost and simple adjustment and operation.

Circuit Operation

The phase-shift oscillator is the simplest circuit found in common transistor circuitry. Two basic changes were made to this circuit. First, R1 was made adjustable (Fig. 6-13) to set the mark frequency (2125 Hz). Secondly, R2 is tapped and an FET placed between the tap and ground.

An FET will conduct as long as the gate-source and gate-drain junctions are not reverse biased. With the gate grounded, the U112 FET exhibits about 500 ohms between source and drain. However, when a positive voltage (greater than 6 volts for the U112) is applied from gate to source, the U112 is pinched-off. In the pinched-off state the resistance from

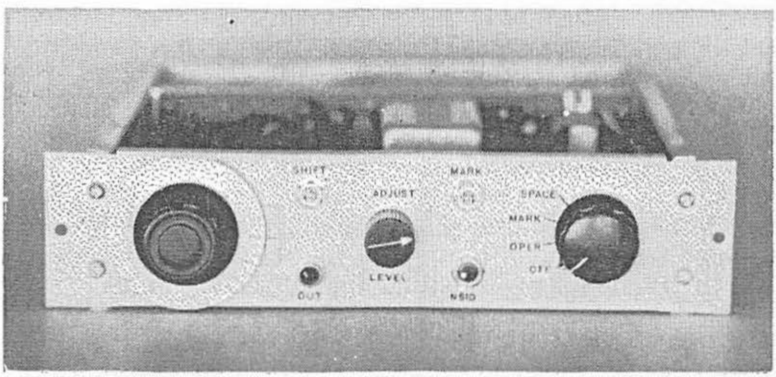


Fig. 6-12. The completed novel AFSK oscillator unit.

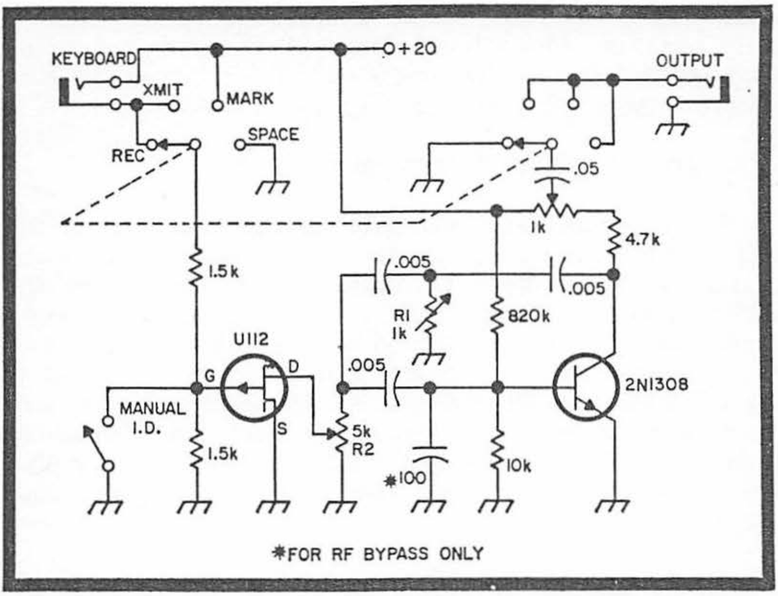


Fig. 6-13. The AFSK phase-shift oscillator used in the solid-state RTTY system. This circuit features equal mark and space output levels, no switching transients, isolation from the keyboard, and a simple shift system—an FET used as a resistor.

source to drain is extremely high and can be considered to be infinite for our purposes.

With positive voltage on the gate of the U112, the phase-shift circuit is unaffected and the mark frequency can be set with R1. When the gate is grounded (positive signal removed) the U112 conducts, placing 500 ohms across a portion of R2 lowering the resistance of this arm of the phase-shift network, and raising the frequency of the oscillator. In this state, the space frequency (2975 Hz), can be set with R2.

As long as a voltage greater than six volts is applied to its gate, the U112 will be pinched-off. This gives complete isolation from any resistance changes in the keyboard—if the divider network is properly designed. The circuit values shown draw a little less than 10 ma through the keyboard contacts to keep them clean, and any changes in keyboard contact resistance are small compared to 3K ohms.

The positive signal for the gate can be derived as shown, or from the printer local loop. In either case be sure to have a small resistor from gate to ground. The input resistance of these devices is so high that a charge on the 10 pf gate-source

capacitance will take a long time to decay (the better part of a second!) unless shunted by a much smaller resistance. The decay has the effect of slurring the markspace transition, and is slow enough to be easily heard. Also, be sure not to exceed the gate-source breakdown voltage, listed at 20 volts maximum for the U112.

Output of the oscillator is several volts peak-to-peak. The fixed resistor in the collector circuit isolates the output load from the phase-shift network. Without this, setting the output pot to the collector end loads the network and reduces the frequency of oscillation. The oscillator as shown is sensitive to supply voltage changes so a regulated supply is necessary.

The terminal unit poses more of a problem. There are several good transistorized circuits available, but they all use 30 volts or less to drive the selector magnet. The selector magnet will not pick up properly unless the change of current versus time is large; this requires a large voltage driving the magnet. Most printers will not print properly with a 20-volt supply, but will with a 100-volt supply (both at 60 ma, of course). A solution to this problem might be to purchase some of the high-voltage transistors available. However, in this case the SCR equivalent of the thyratron commutator was used (Fig. 6-14). This is simply a method of switching a DC load on or off using two SCR's. Essentially, when SCR1 is on, and SCR2 off, point A is grounded and load 1 is activated. When a positive control signal arrives at the gate on SCR2, it turns on, grounding point B. Capacitor C1 has been charged to the supply voltage and grounding point B applies a negative voltage to point A, the anode of SCR1, turning it off. SCR1 will stay off if its gate is less positive than the necessary trigger signal. A signal at the gate of SCR1 will reverse the action. With this commutator to carry the current for the selector

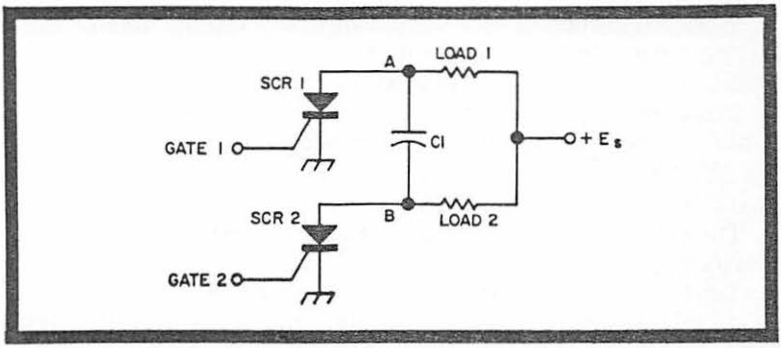


Fig. 6-14. Basic SCR circuit which is used to drive the printer magnet.

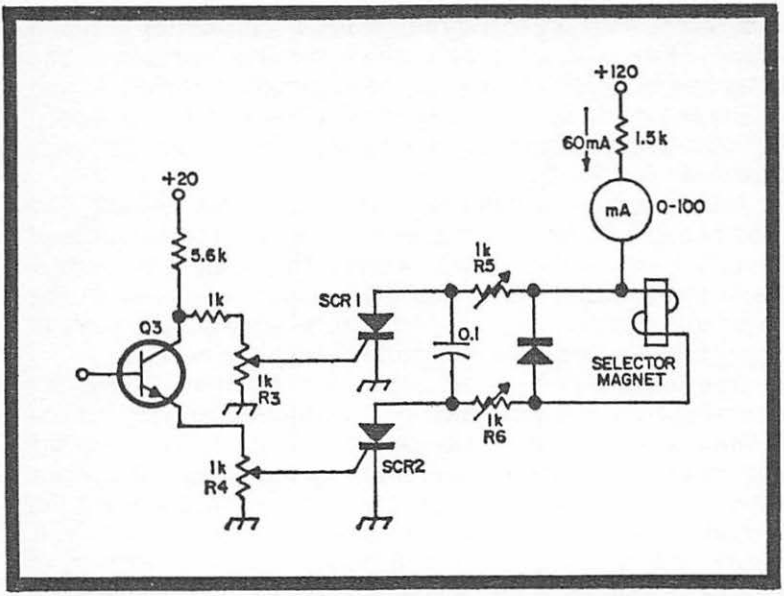


Fig. 6-15. The complete SCR printer-magnet driver circuit with the trigger Q3.

magnet (at any voltage up to several hundred), the only requisite was to fine a suitable triggering circuit.

Fig. 6-15 shows the triggering circuit, along with the SCR circuit. With Q3 off, there is no voltage across R4 and about 3 volts across R3, which can be set so that SCR1 fires. When a positive voltage is applied to the base of Q3, turning it on, about 3 volts appears across R1, which can be set so that SCR2 fires. The voltage across R3 drops when Q3 conducts, removing the gating signal from SCR1. When Q3 is turned off the action reverses again.

With this circuit it is a simple matter to adapt one of the discriminators to drive the switch. Fig. 6-16 shows the whole circuit. Diodes 1 and 2 provide simple limiting. This is adequate for strong signals. For weaker signals, a bandpass amplifier with AGC might be added ahead of this circuit. The rest of the circuit is self-explanatory except for R1. This provides no-signal bias to Q3.

Tunable inductors were used to make adjustment of the discriminator easy.

Tune the system, place the reversing switch to normal and the standby switch to standby. Apply a mark signal (2125 Hz) to the input and a VTVM to the test point (TP). Tune the mark filter for maximum voltage at TP. Switch the input to a space

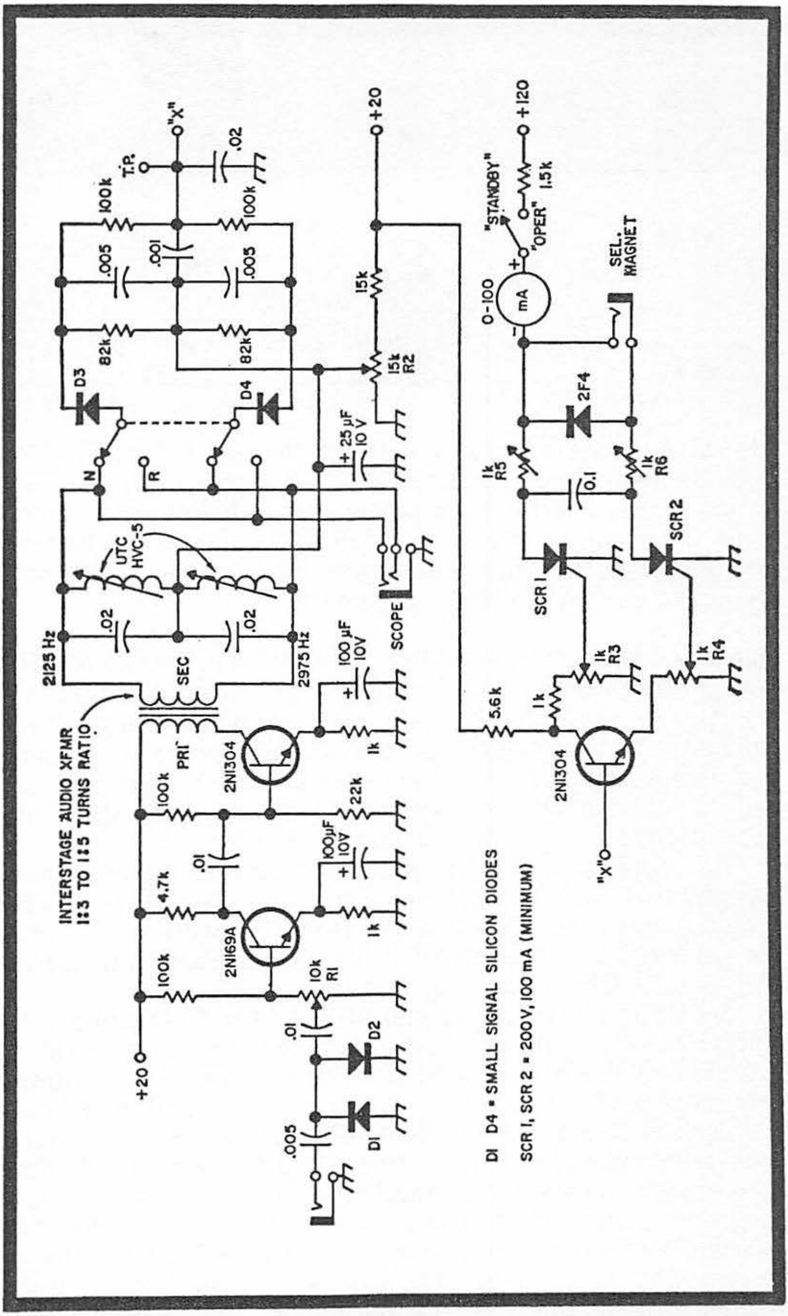


Fig. 6-16. Schematic of the complete solid-state RTTY tuning unit.

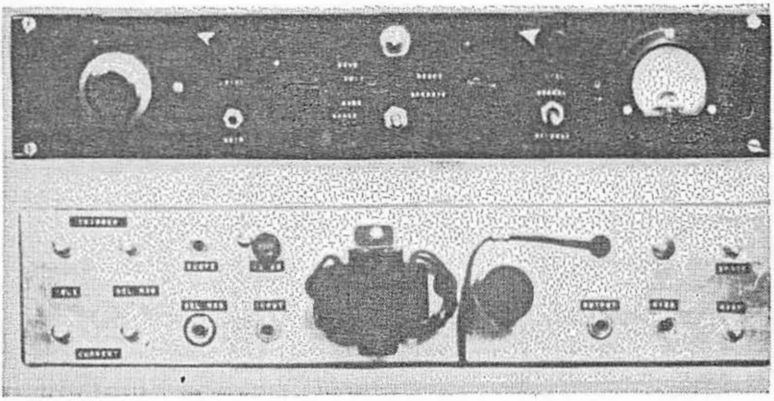


Fig. 6-17. Front and back view of the solid-state RTTY unit. On the front panel, top, the AFSK is to the left, the TU on the right. The ten-turn pot on the extreme left is not used. In the back view, bottom, the TU controls are on the left, AFSK on the right. The power transformer and filter capacitor are mounted in the center.

signal and tune the space filter for minimum voltage at TP. With a large enough signal, this voltage should go negative.

Remove the signal and vary R2 through its range. The voltage at TP should go from zero to some maximum, with a sharp knee around 3 volts. This knee marks saturation of Q3 and should be noted. R2 is set so that the no-signal voltage at TP is midway between zero and the saturation point.

Now apply a mark signal again and adjust the gain so that Q3 saturates. Switch to a space signal and the voltage at TP should go negative. If the voltages from the discriminator are not symmetrical, adjust R2 slightly so that a reasonable signal will saturate Q3 on mark and cut it off on space.

Set R3 and R4 to ground and R8 and R5 to their mid-point. Apply a mark signal and switch the standby switch to operate. Increase R1 slowly until SCR2 fires. This is noted by the jump in current and by the selector magnet pulling in. Adjust R5 for the desired 60 ma. Now switch the input to a space signal. Increase R3 slowly until SCR1 fires. This should be noted both by a change in the current and by the selector magnet dropping out. Adjust R5 for the same current as drawn by the selector magnet. A little playing around with R3 and R4 may be necessary to get the proper switching action from a weak signal. Now tune in a station and listen.

The power supply is not shown. As mentioned, a zener regulated supply is necessary. The high voltage supply uses a

simple half-wave rectifier with RC filtering, the 1500-ohm current limiting resistor is included as part of the power supply. The regulation on this supply is not too important, as long as it will supply the 60 ma and maintain 100 volts or more.

Both the AFSK oscillator and the tuning unit are built into a 3½ inch relay rack panel and recessed channel as shown in the photographs. Operating controls are on the front panel (Fig. 6-17). Frequency and current adjustments are on the back, along with all jacks, the fuse, and power supply components. The AFSK oscillator is built onto a small circuit board attached to the front panel. The tuning unit is built on a similar board mounted parallel to the panel. The photographs show a ten-turn pot on the left (Fig. 6-18). This is not used and the space may be large enough to mount a 1'' scope for tuning, if the desire and funds permit.

On the weaker stations, garbled copy from fading is annoying and the bandpass filter with AGC as mentioned above would be valuable.

A VERSATILE, CRYSTAL-CONTROLLED AFSK GENERATOR

This AFSK generator uses digital IC's. In addition to providing an accurate source of mark and space tones (accurate to less than 1 Hz with tone amplitudes within 1 db of each other) it also contains a microphone preamp. The generator displays no keying transients, and three methods of keying are provided.

It is conceivable that a generator could be designed using a stable 425-Hz source and then selecting the 5th and 7th harmonics to produce the mark and space tones of 2125 and 2975 Hz. However, no practical means of executing this idea

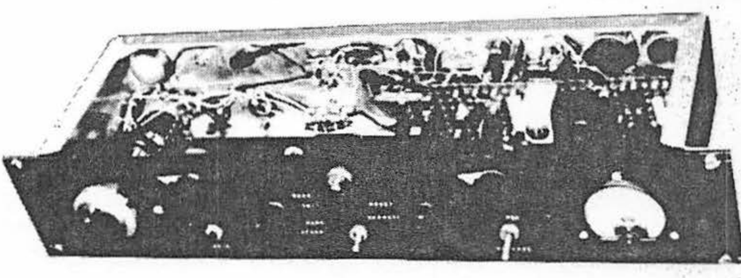


Fig. 6-18. Top view of the solid-state RTTY unit. The tuning unit circuit board is on the right. The FET in the AFSK oscillator is soldered to the pot in the upper left hand corner of the chassis.

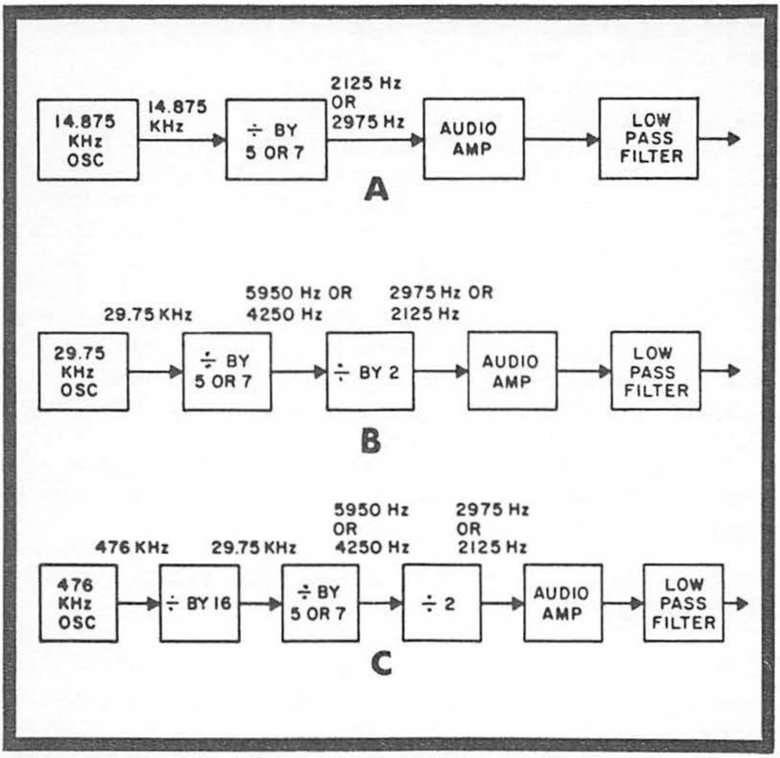


Fig. 6-19. Block diagrams showing evolution of tone generator development.

was apparent until Motorola RTL digital IC's came on the market at a reasonable price. Although the IC can not be used to multiply, they can divide. So, the design concept was reversed (Fig. 6-19):

If two frequencies are each harmonics of another frequency, it follows that they are also sub-harmonics of still another frequency. In this case 2975 and 2125 Hz are the 5th and 7th sub-harmonics of 14.875 Hz. Now, we have arrived at a single signal source to provide both mark and space signals.

Since this device is to be used on 2-meter FM, sufficient audio was required to drive the primary of the usual carbon microphone transformer. An RCA CA3020 IC audio amplifier is adequate for this purpose, and it doubles nicely as a microphone preamp as well. However, there is the problem of transforming the IC divider's square wave output into sine wave tones. This is accomplished with a low pass filter following the audio amplifier, suppressing the square wave's higher harmonics, and producing sine waves.

The original idea is shown in Fig. 6-19A. The new approach presented the problem of passing 2975 Hz through the low pass filter with less than 1 db attenuation, and, at the same time, suppressing 4250 Hz (the second harmonic of 2125 Hz) to at least 40 db down. However, this turned out not to be a problem since, according to Fourier, there are no even harmonics generated by a symmetrical square wave—just odd ones. So, if the input frequency of 14,875 Hz is doubled to 29.75 kHz, we can then add a divide-by-two stage after the divide-by-five or -seven stage. Now, the lowest frequency to be concerned about is the third harmonic of 2125 Hz, or 6375 Hz. It is taken care of by the simple 5th order Chebyshev filter. (Fig. 6-19B).

One way to obtain the basic 29.75 kHz signal source would be to use two cross-coupled NOR gates in an astable multivibrator. But these could be trouble-makers, since the oscillation frequency can be affected by supply voltage and temperature variations. Another possibility would be by use of a unijunction transistor relaxation oscillator. It is relatively immune to voltage changes and can be temperature compensated.

An even better approach is by use of a crystal oscillator. Crystals at 29.75 kHz are expensive. But a surplus FT241 crystal that is the 16th harmonic of 29.75 kHz is easily and cheaply obtained. The crystal used is marked Channel 57, 25.7 Mcs, cut for 475.925925 kHz. So, the final configuration of Fig. 6-19C evolved.

The 476-kHz oscillator precedes a divide-by-sixteen stage. Then comes a divide-by-five or -seven stage, followed by a divide-by-two stage. The latter feeds the audio amplifier, followed by a low pass filter. The result pure, stable, crystal-controlled mark and space tones.

Circuit Details

1. **476 kHz square wave generator**—NOR gates G1A (Fig. 6-20) form an astable multivibrator with a free running frequency slightly below 476 kHz. Potentiometer R1 controls that frequency. When crystal Y1 is hung across the two inputs to the gates, it tries to synchronize the frequency to its own resonant frequency. Synchronization, of course, depends upon the free-running frequency, therefore, as R1 is varied, so is the frequency, by a slight amount.

2. **Divide-by-sixteen stage**—A simple binary counter is assembled by the J-K flip-flops. FF1A, FF2A, FF3A and FF4A. It divides the input frequency by 16. Therefore, if the input is 476 kHz, the output will be 29.75 kHz. Although the

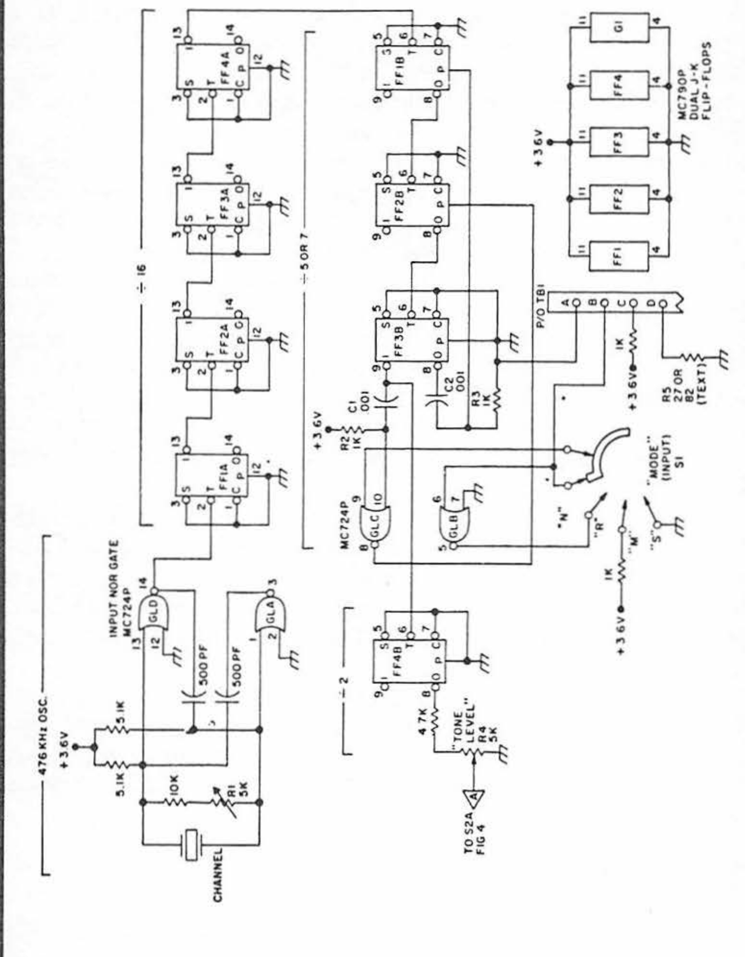


Fig. 6-20. Digital mark-space tone generator schematic

more observant may note the counter actually functions backwards, this configuration was used for wiring convenience. Either way, forward or backwards, it takes 16 cycles of input frequency to get one cycle of output frequency.

3. Divide-by-five-or-seven stage—The heart of the AFSK generator is the divide-by-five-or-seven stage. Before detailing how this circuit is used, we should first review the Motorola RTL logic circuits. First, the simple NOR gate: if any input is high, the output is low. If we use positive logic, (i.e., high voltage is 1, low voltage is 0), a truth table for a 2-input NOR gate would look like this:

Inputs		**Output**
A	B	
0	0	1
1	0	0
0	1	0
1	1	0

The J-K flip-flop is more complex. Basically, it has three inputs and two outputs. Set, clear and trigger are the terms for inputs; outputs are 0 and 1. In the 0 state, the flip-flops 0 output is high and the 1 output low; in the 1 state, the 0 output is low and the 1 output high. To place the flip-flop in either of these states, various combinations of the inputs are used.

If both the set and clear inputs are at a logic 1 level, and a 1-level pulse is applied to the trigger input, the flip-flop changes state, or reverse itself. If the set input is at a logic 1 level, and the clear input at a logic 0 level, and a logic 1 pulse is applied to the trigger input, the flip-flop goes into the 1 state. If the set input is at a logic 0 level and the clear input at a logic 1 level, a logic 1-level pulse applied to the trigger input will induce the flip-flop into the 0 state.

Referring to Fig. 6-20 note that the Motorola RTL J-K flip-flops operate slightly differently. The S, T and C inputs as well as the 0 and 1 outputs have little circles after them, denoting inverters. They invert the logic into and out of the flip-flop, thus converting a 0 to a 1 and a 1 to a 0. So, the truth table for the Motorola J-K flip-flop would look like this:

Inputs *		**Outputs** * *
S	C	1 0 State
0	0	Changes State
1	0	1 0 0
0	1	0 1 1
1	1	No State Change

* Before Negative Pulse to T.
* * After Negative Pulse to T.
So, if the flip-flop is in the 1 state, the output from the 1 output (after it goes through the inverter) is low, or a 0 logic level.

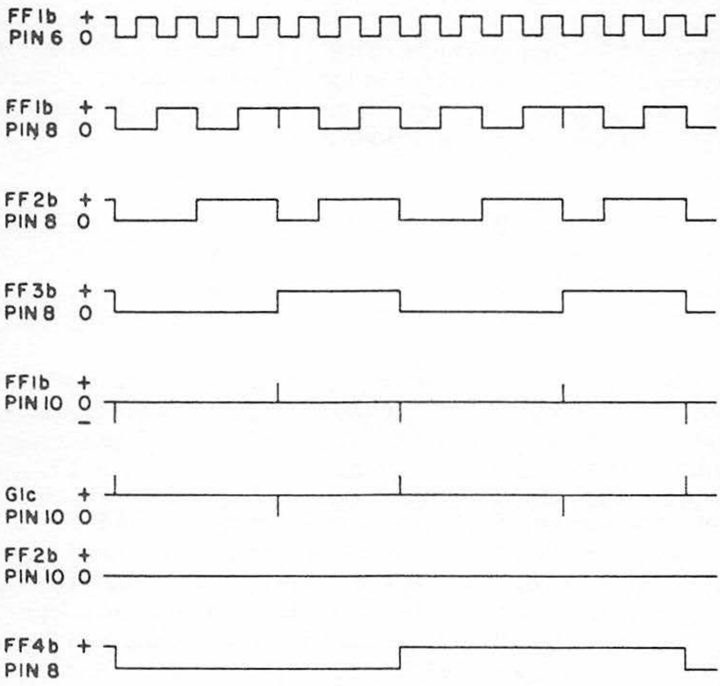


Fig. 6-21. Mark condition waveforms.

The Motorola RTL J-K flip-flop has an extra input called Direct Clear. This may be confusing since, if a positive pulse is applied to it, the flip-flop goes to the 1 state, regardless of the inputs to the other three input terminals. The Fairchild RTL series also uses this extra input, but terms it Preset, which more accurately describes its function.

The divide-by-five-or-seven stage is formed by flip-flops FF1B, FF2B, FF3B and NOR gate G1C. Referring to the waveform diagram, Fig. 6-21, observe how the divide-by-seven state operates. Basically, these flip-flops form a divide-by-eight counter. However, when the fourth input cycle is received, FF1B and FF2B go into the 0 state, with FF3B going to 1 state. The positive going output from pin 8 of FF3B goes into the RC network R3C2, which is a differentiator. It takes in a square wave and produces a spike out, maintaining polarity.

The positive spike out of the network feeds the Preset input of FF1B, placing it in the 1 state. Thus we have deceived the counter into believing it has received an extra pulse. From this point on, the counter again counts normally. This is the counting sequence:

000
001
010
011
101
110
111

Note we have skipped the 100 state and it takes only seven input cycles to get one output cycles.

Pin 9 of FF3B also has a differentiator following it, R2C1. When FF3B goes into the 1 state, it produces a negative spike out. This is fed to NOR gate G1C. However, since the other

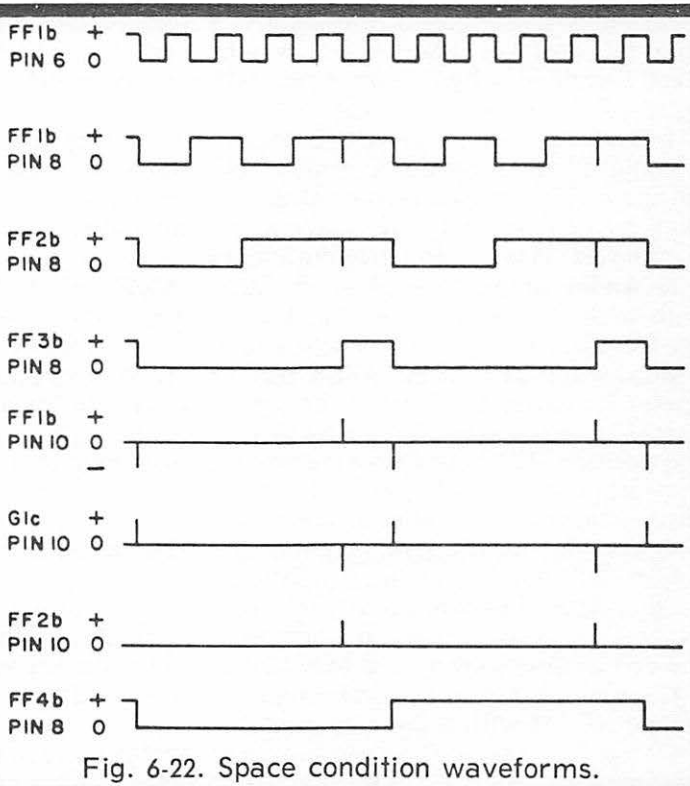


Fig. 6-22. Space condition waveforms.

input to G1C is high during the divide-by-seven sequence, the output is always low. During the divide-by-five sequence, pin 9 of G1C is at a low level. Pin 10 has returned to +3.6 volts through R2, thus the output is still low.

Examining waveform drawing Fig. 6-22, we note that when FF3B goes into the 1 state, the negative pulse from R2C1 is fed into G1C. Since pin 9 of G1C is already low, when pin 10 goes low, a positive voltage appears at the output. Therefore in this divide-by-five state, G1C inverts the input to pin 10. The output from G1C is fed to the Preset input of FF2B. Now both FF1B and FF2B are placed in the 1 state, and the counter believes it has received 3 extra pulses. The counting sequence appears like this:

000
001
010
011
111

This time we have skipped the 100, 101 and 110 states and it requires only five-cycles at the input to get one cycle at the output. It is the level at pin 9 of G1C which determines whether the counter divides by five or seven. G1B simply inverts the input level for reversed keying.

4. **Divide-by-two stage**—FF4B forms a simple divide-by-two stage. It takes the non-symmetrical square wave output from the divide-by-five-or-seven stage at 4250 Hz, or 5950 Hz, and produces a symmetrical square wave output of either 2125 Hz or 2975 Hz. This is then fed to the audio amplifier.

5. **Audio amplifier**—IC1, and RCA CA3020 integrated circuit audio amplifier (see Fig. 6-23), accepts the square wave from the divide-by-two stage and amplifies it when the function switch S2 is in the AFSK position. Tone level potentiometer R4 varies the level of the square wave into the amplifier (Fig. 6-24). When switch S2 is in either the PTT or the voice position, IC1 functions as a microphone preamplifier. The microphone is fed to pin 10 of IC1 which is the base of an emitter-follower. This follower's emitter is pin 1 of IC1 and connects to R6, the Mic level potentiometer. The audio is then fed back into IC1 for further amplification.

The output impedance of IC1 is about 130 ohms. Matching was achieved by using two transformers, T1 and T2, with their voice-coil windings connected back-to-back. T1 is an Argonne AR-176 with a 125-ohm centertapped primary. T2 is an Argonne AR-164 with a 500-ohm primary.

6. **Output Network**—There are three separate stages in the output network. R8 and R9 form a 500- to 430-ohm minimum

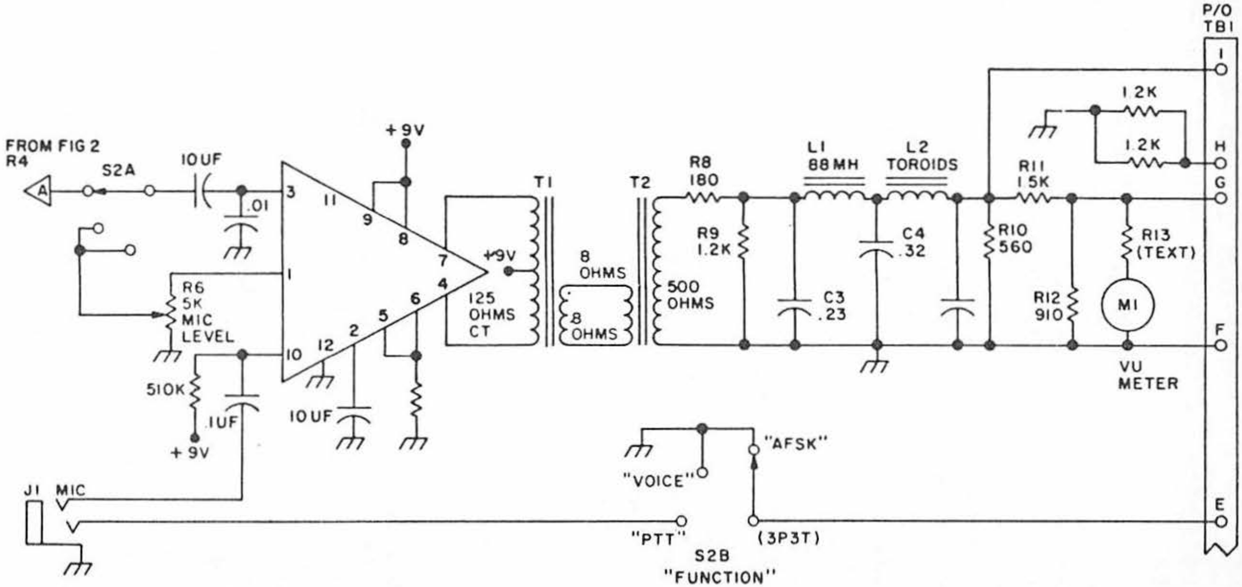


Fig. 6-23. Audio amplifier and output network schematic

loss pad. C3, L1, C4, L2 and C5 form a five-section Chebyshev low-pass filter. R10, R11 and R12 form a 430- to 600-ohm 16-db pad. The requirements for the low-pass filter led to this unique arrangement.

Since standard available inductors were to be used, the choice was confined to 11, 22, 44 or 88 mh. The filter had to be 40 db down at 6375 Hz. A large pad appeared to be necessary to isolate the low-pass filter from the load since the primary impedances of various carbon microphone transformers might differ considerably. Since it was desirable to lump most of the loss on the output side of the filter, it should have an impedance near 500 ohms to minimize the loss in the input matching pad. Using 22 mh in the design equations for the low-pass filter resulted in an impedance closest to 500 ohms. It varies directly with the 40 db down frequency. Since the minimum 40 db frequency is 6375 Hz, that figure and 22 mh in the equations came out to 433.5 ohms.

R8 and R9 match the 500-ohm output of the audio amplifier to the 433.5-ohm input of the low-pass filter. Insertion loss is about 3.28 db for this minimum loss pad. The low-pass filter is a five-section Chebyshev which means its skirt is fairly sharp, but it does have some ripple in the passband. It is about 1 db and its 3 db point is above 3500 Hz. At 6375 Hz, it is 40 db down, and much more so, of course, at 8925 Hz, the third harmonic of 2975 Hz.

In computing loss through the low-pass isolation pad, it was determined that 4 to 5 milliwatts should be sufficient to drive most carbon microphone inputs. This figures out to about 20 db loss between the audio amplifier and output. The input of the low-pass filter already shows a 3.28 db loss through the minimum loss pad. Therefore, about 16 db would be needed for the output pad. R10, R11 and R12 form this pad which also matches the output impedance of the low-pass filter to 600 ohms. Using standard value resistors, the loss actually comes out to 15.53 db. This produces a total loss of 18.81 db in the pads and between 0 and 1 db in the low-pass filter. Shorting the output, the filter sees 407.8 ohms; with an open circuit, it sees 454.4 ohms. The maximum variation between open and short is less than 6 percent, an excellent isolation.

Meter M1 is an illuminated miniature VU meter (Fig. 6-24). This is a B-scale VU, where 0 VU is 1.228 volts rms, when used with the precision resistor supplied with it. This corresponds to +4 dbm across a 600-ohm line, or about 2.5 mw. The external multiplier resistor can be changed to display a 0 VU reading for other levels if desired.

Fig. 6-24. The completed unit, housed in an aluminum box covered with wood-grained contact paper.

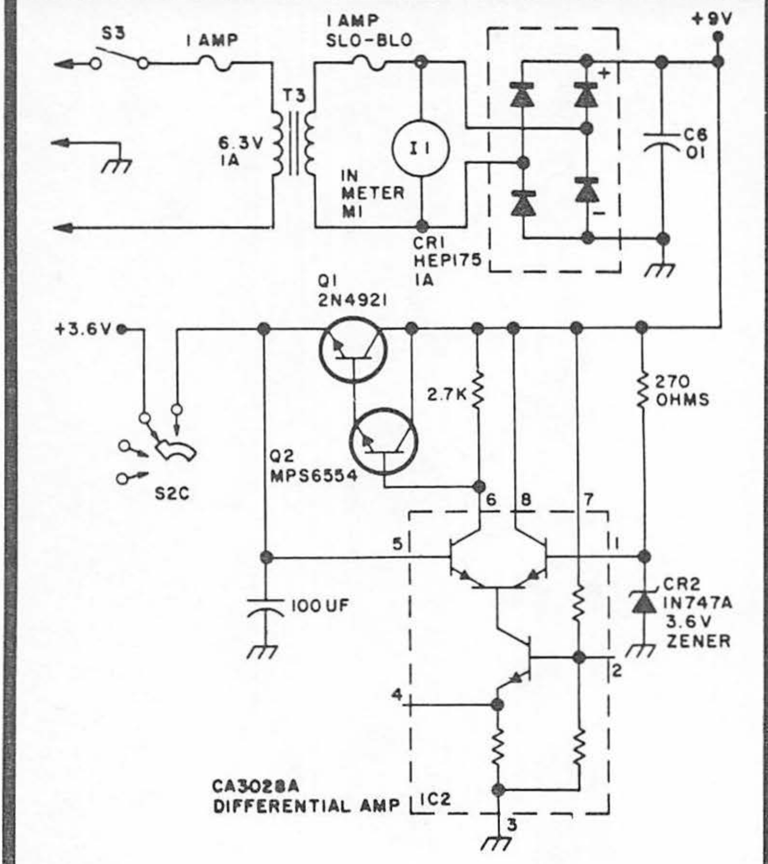


Fig. 6-25. Power supply schematic.

7. **Power supply**—The power supply (Fig. 6-25) is fused on the input side. T3 is a 6.3 volt, I amp filament transformer. It is also fused on the secondary since a short probably would not take out the primary fuse. A slow-blow fuse is required because of the large peak currents drawn due to the large value of C6.

I1 is part of the VU meter, M1. CR1 is a full-wave bridge rectifier in a single neat package. C6 is a 10,000 mfd (.01 farad) 25 volt capacitor (Fig. 6-26). Depending on load, the output is between 8 and 9 volts. Most of the voltage drop under load appears to occur in the transformer winding. Therefore, a huskier transformer should provide better regulation. Output powers the audio amplifier and the +3.6 regulator.

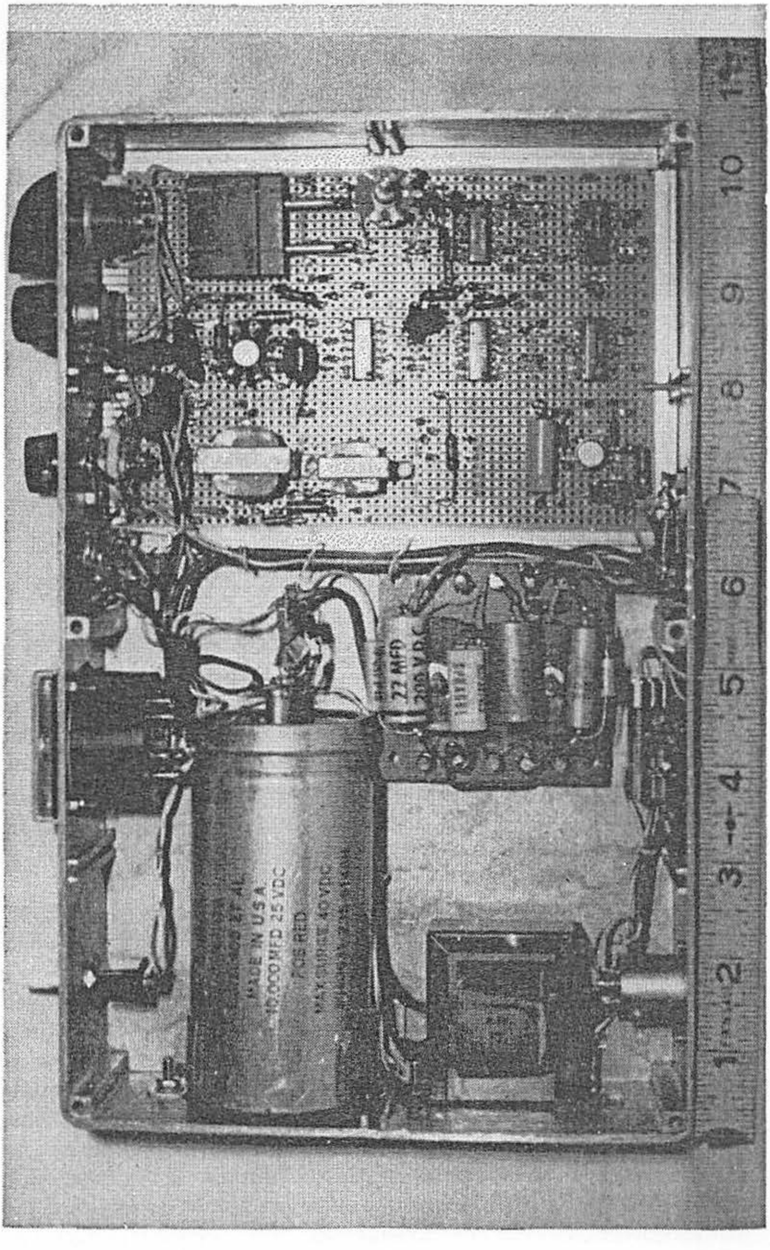


Fig. 6-26. Interior bottom view, showing plugboard. The bridge rectifier was mounted directly on the filter capacitor to avoid ground loop problems. Note the heat sink on IC1.

The regulator is required to supply +3.6 volts to the digital IC's. The voltage holds steady under varying loads and eliminates any residue ripples that might leak through C6. Diode CR2 is a 3.6-volt zener. This regulated voltage is fed to pin 1 of IC2, an RCA CA3028A. IC2 is a high gain differential amplifier. Its function is to compare the reference input on pin 1 to the regulator output on pin 5. The output is then fed to the base of Q2, connected to Q1 in a Darlington circuit. So, if the output is higher than the reference, the output on pin 6 of IC2 is lowered. This in turn reduces the base current to Q2 which lowers the base current to Q1. This drops the output voltage on the emitter of Q1. The reverse process occurs if the output is lower than the reference.

Although fair regulation might have been obtained with zener diodes, there was doubt about their performance at low voltages. This regulator maintains output voltage changes to less than .1 volt and there are no ripple or transient problems. Even this small voltage variation might be avoided by using a larger primary power transformer.

Construction

Although the prototype was developed on a Vector 3477 DIPlugboard, constructors are advised to use standard unclad perfboard both for ease of construction as well as to minimize coupling between the close-spaced etched leads on the DIPlugboard. This type of board also is susceptible to ground loops. It is advisable, even using standard perf-board, to use a common ground point for the bottom ends of R4, R6 and R7 and the ground ends of the bypass capacitors on pins 2 and 3 of IC1.

Particular care must be observed in shielding, since harmonic-rich RF square-waves are being generated. Use of an RF tight metal box is essential.

Fig. 6-27 should simplify identification in IC and transistor lead basing. Looking at the MC700P ICs from the bottom with the notch on the left, Pin 1 is on the left end of the top row of pins. The remainder of the pins are numbered consecutively clockwise. Pin 14 being on the left end of the bottom row. Viewing the CA3000 ICs from the bottom, note the little tab on the case. It is adjacent to the highest numbered pin. Pin 12 of the CA3020 and Pin 8 of the CA3028A. These pins are also numbered clockwise. The 2N4921, Q1, may be confusing. The case is rectangular plastic with three leads on the bottom and a copper plate on one side. There is a hole through it. Viewing from the bottom, with the copper plate up, the base lead is to the left, the collector in the middle, the emitter on the right.

Alignment

Alignment is simple. Without connecting to the rig, jumper pin G to Pin H on TB1. This places a 600 ohm load on the unit. Place the function switch S2 to AFSK. Pots R4 and R6 should be at minimum. Mode switch, S1, should be in either the M (mark) or S (space) position. When turning on the power, it should light. At this point, check for proper supply voltages. Using the original multiplier resistor that came with M1 as R13, adjust R4 until M1 reads 0 VU. This interprets to about 1¼ volts across the output.

A frequency counter, if available, should be used for alignment. Check the output of FF1A at Pin 14. Adjust R1 until the output is exactly 238 kHz. If no counter is available, accurate mark or space tones, either off the air from an obliging ham or from a tape standard can be used. Place S1 to the tone you are aligning against, M or S, and connect the generator output to a scope. Use the output at Pin 1 on TP1 (Fig. 6-28), since thisis high impedance and compatible with most scopes. Put the standard tone on the other scope input axis and adjust R1 for a 1+1 pattern on the screen. When one tone is adjusted, the other is automatically on frequency. A third method of

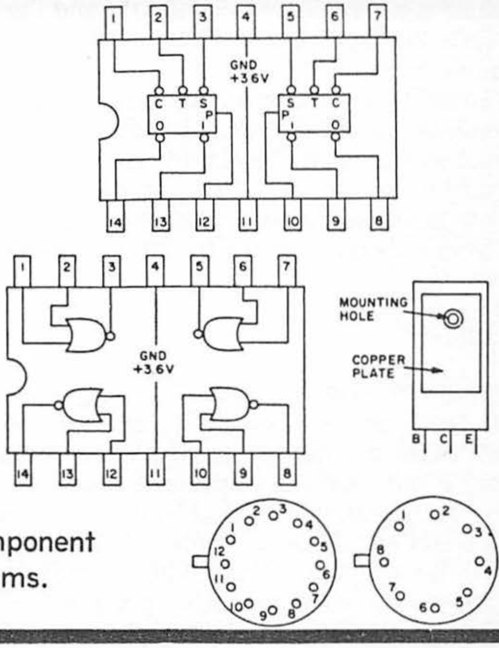


Fig. 6-27. Component basing diagrams.

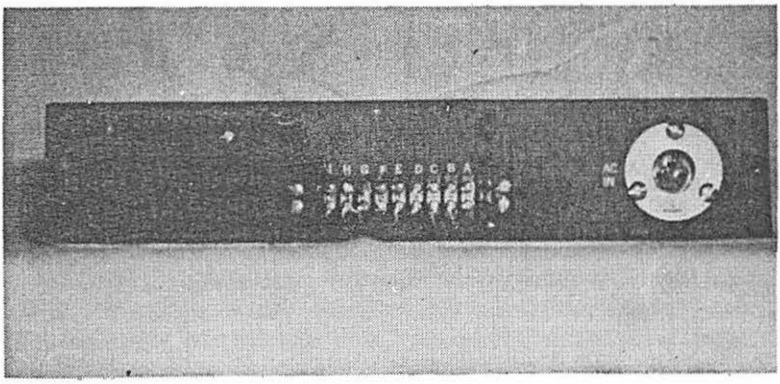


Fig. 6-28. Rear view of unit.

alignment is to use a 1430-kHz broadcast frequency. (WNJR in New Jersey for the East Coast). If an antenna is attached to the bc receiver and placed close to the 476-kHz square-wave generator, you should hear a beat note. This is the third harmonic of the generator and the 1430-kHz signal. With the unit placed in the mark condition, connect the output to one axis of a scope. The audio of the bc receiver goes to the other axis. Adjust as described earlier for a 1:1 pattern between the beat note and the mark tone. Actually, the square-wave generator is producing 475.95840 kHz and the mark and space tones are 2124.8143 Hz and 2974.7400 Hz, respectively. The maximum error is 0.26 Hz!

Tuning of the low-pass filter is a simple matter. No test equipment is required. The only value to be adjusted is that of C4. Start with a value of 0.22 mfd. Then add capacity in steps of 0.001 mfd until the mark and space tones are within 1 db of each other as indicated on M1. Now disconnect the jumper from terminals G and H of TB1 and you are ready to hook it up to your rig.

Operation

Three different methods of keying are provided (Fig. 6-29). If the local loop supply is grounded and well-filtered, break it at the ground point and connect to terminal D of TB1. The loop ground goes to terminal A. Jumper terminals B to D. In this configuration the local loop runs through R5. For a 60 ma loop, R5 should be 27 ohms. With a 20 ma loop, R5 is 82 ohms. In the mark condition, with current flowing through the loop, about +1.6 volts appear on pin 9 of G1C with S1 in the N (normal) position. This serves to inhibit G1B and causes the

unit to generate a mark tone. When the loop is open, or in the space condition, 0 volts appear on pin 9 of G1C, enabling G1C, and the unit generates the space tone.

Polar relay keying is also provided by connecting the common to terminal B, the mark contact to terminal C and the space contact to terminal A. In this arrangement, G1C receives +3.6 volts through a 1K resistor in the mark condition and a ground during space.

Direct keyboard keying can be used by connecting terminal B to terminal C, and connecting the keyboard between terminals A and B. Although keying will be inverted, it can be corrected by placing S1 in the R (reverse) position. The keyboard could be connected between terminals B and C for normal keying, but this would result in the input line on terminal B being left open during space. This could produce hum.

It is assumed that S1 is in the N (normal) position in the preceding instructions. In the R (reverse) position, pin 9 of G1C is no longer fed from terminal B of TB1. It is now connected to the output of B1B, which takes the input from terminal B and inverts it. Therefore, the keying is inverted. In the M (mark) position, S1 connects the input of G1C to +3.6 volts through a 1K resistor; in the S (space) position it grounds the input to G1C.

To apply the generator's audio output to the transmitter, any voltages connected to the primary of the carbon microphone transformer should be removed. Terminal G of

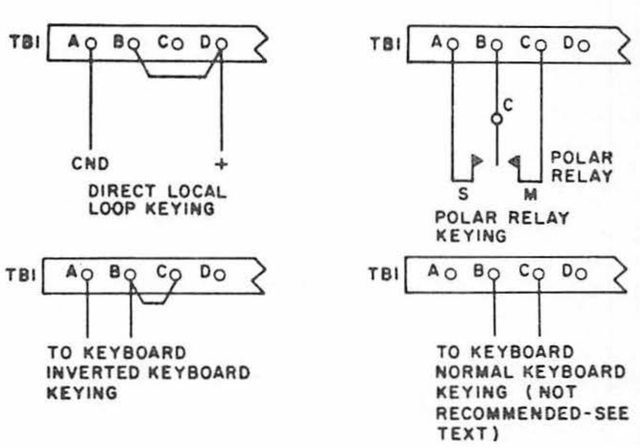


Fig. 6-29. Keying connection diagrams.

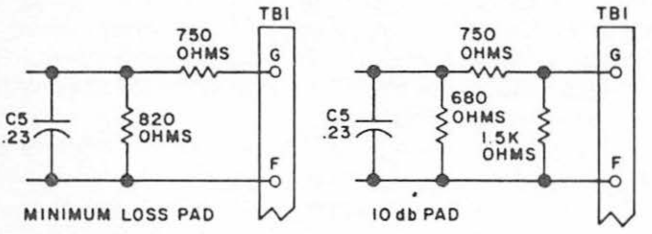


Fig. 6-30. Alternate output pad diagrams.

TB1 should then be connected to the high side of the transformer and the low side to terminal F. Connect the PTT line to terminal E. With the transmitter in the standby condition, and the generator operating, placing S2 in the AFSK position should activate the transmitter. Advance the tone level pot to achieve approximately 100 percent modulation. Then select a value of R13 that displays 0 on the VU meter. Place S2 in the voice position and advance the mic level pot to produce about -3 VU while speaking normally into the microphone.

In the event the audio levels are low for your particular transmitter, it might be due to a mismatch between the 600-ohm output of the generator and the microphone transformer. This could be remedied by connecting a 600-ohm center-tapped transformer to the generator's output, and then hooking the microphone transformer between the low side and the center tap. We then have an autotransformer with a 600: 150-ohm ratio. Or the carbon microphone transformer could be replaced with another unit with a 600-ohm primary.

If more output is desired, and it is evident you have a 600-ohm load for the generator, the 16-db matching pad could be replaced with a 430:600-ohm minimum loss pad as shown in Fig. 6-30. This pad has about a 5.2 db loss, and the low pass filter sees slightly more than 0.5 percent error in termination when the output of the pad is terminated in 600 ohms. This should produce an added 10 db of gain, provided the unit is terminated at 600 ohms. Or the 16-db pad could be replaced with an alternative 10-db pad. The gain in this instance is about 6 db in the output, but care in the termination value must be observed. Going from short to open circuit makes the low

pass filter see a variation of about 20 percent from its design impedance. This is much greater than the 6 percent variation with the 16-db pad.

Still another method of obtaining more gain is to short out R7. This raises the current drawn by the amplifier and may increase distortion. Or Pin 11 or IC1 could be connected to the +9 volt line through a 1.5K resistor. However, these alternatives may lead to amplifier instability, since the 3 db point of this little IC is about 8 MHz with a resistive load. It is usable up into the VHF range with tuned loads.

Various modes of keying were tried in on-the-air testing of this unit. While all appeared to perform normally and, as anticipated, scope patterns and analyses at the receiving ends indicated that the polar relay method produced the most accurate keying characteristics. If a relay is used, be sure that it is a well adjusted polar relay. This unit will follow keying up to near a 4 kHz rate. Any contact bounce will be faithfully reproduced by the unit. In all modes, there was favorable comment on the purity of the tones as well as their accuracy. Even the most exotic commercial counter was only able to detect a 1 Hz variation on either mark or space.

Although there is an evident mismatch between the audio unit and the carbon mic input transformer of the transmitter, there was no evidence of insufficient drive to the audio amplifier. No appreciable improvement was obtained using various external transformers, although any arrangement which can provide less than a 2 or 3 to 1 mismatch is desirable. Although the microphone is not used on the RTTY frequency, the same rig is switched to other 2-meter frequencies, so the unit can be left connected to the transmitter at all times, and the microphone is available for use whenever desired.

RCA has recently come out with the CA3020A and the CA3028B, newer models of the linear IC's used in this unit. No circuit modification is required to use these later versions, and almost 3 db more gain is available in the CA3020A than in the CA3020.

Conclusion

As one operator observed after listening to this device on the air and learning of its design and construction: "That is a complicated way to get a couple of simple tones." And so it is. However, for the experimenter who wishes to increase his familiarity with IC's and develop new techniques in obtaining standard results, working with these devices is both challenging and rewarding.

CHAPTER 7

Interconnections and Control Circuits

One of the first problems encountered by newcomers to RTTY operations is the necessity of providing a DC loop circuit and a method of tying the various RTTY units together. The newcomer, naturally, will possess a printer, a receiving converter and his communications receiver.

RTTY INTERCONNECTIONS UNIT

There are many ways of interconnecting these units into a workable system, and here we will describe one simple way of accomplishing the job. As you progress further into RTTY, you no doubt, will outgrow this unit, but the primary purpose of this article is to get the newcomer started as easily as possible.

For example, assume you are using a Model 15 (TG-7B) printer, a TG-11 perforator, a Transmitter-Distributor and an audio-type converter.

A 110 volt DC power supply is needed to supply power for the printer magnets. An interconnections unit is necessary to allow transmit-receive switching, to key the FSK vfo unit, and to allow easy connection and disconnection of auxiliary equipment.

The schematic (Fig. 7-1) shows an interconnections unit which accomplishes these functions.

If your audio type converter has its own polar relay, the output of the converter plugs into J4. This will key the 60 ma DC loop and any other units plugged into the other jacks.

Incidentally, these jacks are of the closed-circuit type and are insulated from the chassis.

K1, the polar relay, is used to frequency shift key a Viking-2 vfo keyer. SW1 is used to provide mark-high or mark-low without retuning the receiver. K2, a SPST, 110-volt AC relay, is used to short out the converter during the transmission period. Voltage for this relay coil is obtained from the rear of the Viking-2. R3 and R4 are metering shunts. Control R2 is used to set the loop current as operating conditions determine change

to maintain the necessary 60 ma. The red connector (selector magnets) plugs into J1, the black connector (keyboard) plugs into J2. These are the two lines from the printer. The transmitter-distributor plugs into J3.

The entire unit is built on a 8" x 6" x 4" chassis. The power supply used is a surplus RA-87 unit, but any good DC supply capable of delivering 100 ma will suffice.

Parts List

K1—W.E. 215-A or 255-A polar relay with socket.
K2—SPST 110v AC normally open relay.
J1, J2, J3 and J4—Closed circuit, insulated phone jacks.
SW1—SPST toggle switch.
SW2—DPDT toggle switch.
M1—75 milliampere meter.
R1—2,000 ohms, 10 watts.
R2—5,000 ohm potentiometer, 25 watts.
R3 and R4—20 ohms, 1 watt.

CONTROL CIRCUIT FOR RTTY

The circuit shown in Fig. 7-2 includes all the operating features that are usually desirable. The addition of the take-over key provides for taking over when the copy starts piling up at the end of the line, is overlining, or the machine is in upper case when it should be in lower. If any of these con-

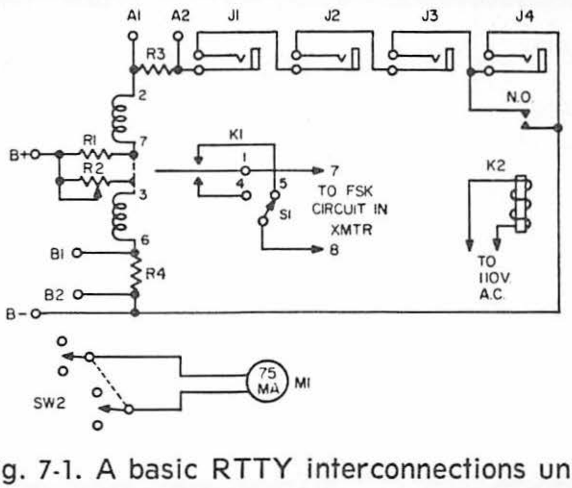


Fig. 7-1. A basic RTTY interconnections unit.

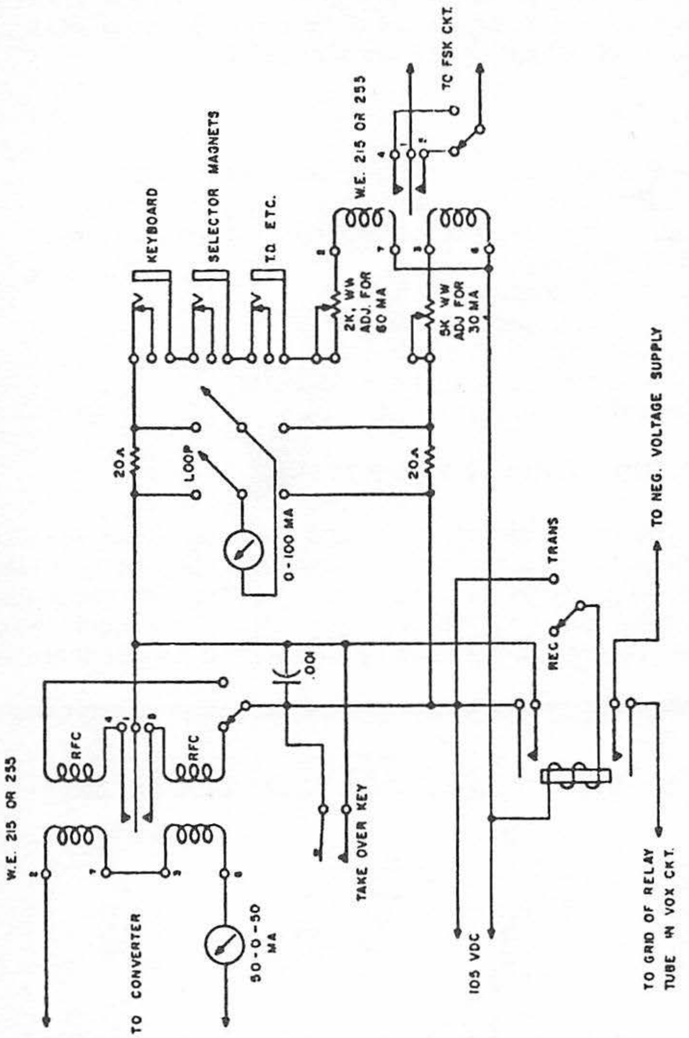


Fig. 7-2. Control unit.

ditions exist, merely operate the take-over key, then operate the proper keyboard keys to get the machine into condition to receive copy. The take-over key does not put the transmitter on the air.

Another feature of this control circuit, that is of interest to those who want to control a transmitter using the VOX circuit, the applying of a negative voltage to the grid of the relay tube in the VOX circuit to provide the same control over the station they now have using VOX. Merely wrap a wire around the pin to the grid of the relay tube in the VOX circuit, and the entire station is controlled by one switch.

SIMPLE CURRENT CONTROLLER

Frequently the experimenting ham finds need of a device to control current in a circuit, rather than voltage. One example of such a need is in a RTTY local loop, where current should be maintained at 60 ma (for a Model 15) even though line voltage, and, as a result, the DC supply voltage available for the loop may vary. Another example is the bleeder resistor of a power supply, which must always draw a minimum current but which is only wasting power if more than that minimum is drawn.

The conventional approach to this situation is to use a voltage much higher than desired across the load, and drop it through a high-valued resistor. For instance, RTTY circuits often use a 125-volt supply and a 2000-ohm resistor, so that 60 ma can flow under short-circuit load. Small voltage changes then result in little current change.

In the case of the bleeder resistor, the resistor value is merely figured so as to draw minimum permissible current with the minimum voltage expected. As voltage rises, so does bleeder current, but we don't worry too much about the wasted power.

Tetrode Characteristics

However, because of the unusual characteristics of tetrode, pentode, and beam-power tubes, it is simple to build a true constant-current generator, or controller, which can be set for any desired amount of current and which will maintain current flow very close to that value regardless of the voltage applied.

Some extra voltage is still needed to operate the tube, but it is usually much less than needed to assure reasonably-constant current under voltage variations, were the conventional resistor hookup employed.

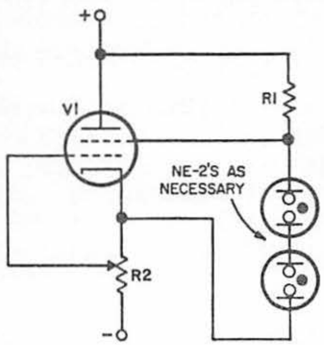


Fig. 7-3. A simple current controller.

The controller works because a tube with a screen grid will, of both screen and control grid voltages are fixed, maintain a constant current flow regardless of plate-to-cathode voltage (except at extremes of operating characteristics). Actually, the current isn't absolutely constant—but if, instead of holding control-grid voltage fixed, we obtain it by means of cathode bias, so that the bias increases as plate current does, then the cathode current will remain virtually constant throughout the tube's operating range.

The circuit is shown in the schematic (Fig. 7-3). No parts values are given because they will depend entirely on the individual application. The tube can be any screen-grid type which will pass the desired amount of current; in general, TV horizontal-output tubes seem to work best as their effective amplification is high. However, the 6V6 is also excellent.

The screen voltage should be chosen to allow the desired current to pass with the grid voltage placed about halfway between zero and cutoff. It should be regulated by a string of NE-2's or VR tubes as shown so that it will not vary with the current. R1 is simply a dropping resistor and should be chosen so that the NE-2's all light under operating conditions.

The value of R2 will determine the range of current control possible. Its maximum setting should be such that cutoff voltage for V1 will be developed by the minimum current desired. Then with the arm toward V1's cathode, current will increase, and with maximum resistance in the circuit, current will be at a minimum.

Circuit Operation

To see just how this works, let us plug in a few numbers. Let us assume we are using a 6V6 for V1, and are holding its

screen voltage at 250 volts. Furthermore, we want to have 20 ma current flowing in the external circuit.

A look at the curves for the 6V6 shows that with a screen voltage of 250 volts, and the same voltage on the plate, a grid-to-cathode bias of 20 volts will allow 20 ma of current to flow. This 20 ma will develop the required 20 volts across a 1000-ohm cathode resistor. So we use a 1000-ohm unit for R2 and bring the grid back to the lower end (with a large pot, we set it for 1000 ohms between grid and cathode).

Now if the load should attempt to make current increase to say 25 ma, the grid bias voltage would also increase to 25, reducing current flow to about 9 ma. Of course, as the current flow dropped, the grid bias would drop accordingly; so that, as current passed 20 ma going down, the grid bias would be back at our 20-volt starting point. The practical effect would be that the current never got a chance to change at all.

If the voltage applied to the current-control and load together were to increase so that there were 400 volts across the 6V6 instead of 250, the 20-volt bias would still hold current to 21 ma. However, 20 ma of current would increase bias to 21 volts, which would result in current dropping to about 18 ma, and, as before, it would stop before, dropping so low. The stop this time wouldn't be right at 20 ma, but would occur at approximately 20.05 ma—which is fairly close control.

Should we want to deliberately increase the current from 20 to say 40 ma, the tube curves tell us that grid bias should be 13½ volts. Ohm's Law tells us that about 338 ohms is the size resistor needed to develop 13½ volts with 40 ma flowing, so that's the setting for R2.

Now an increase in load current from 40 to 60 ma would give us a bias increase to 21¼ volts, which would in turn reduce current to about 20 ma, and on the way down things would lock up at 40 ma where they started, just as before.

Thus, by making R2 adjustable, we can dial the amount of current we want to flow in the circuit, and the controller keeps that current constant.

Should supply voltage drop so low that the screen voltage of V1 drops out of regulation, the gadget fails. This can be overcome by supplying the screen from a separate source, because plate voltage can be allowed to drop far below the screen value before the device stops working. However, in many applications the major problem is a change in current drawn by the load.

Used as a bleeder resistor, this circuit will draw only the amount of current it is designed to pull no matter how high the

supply voltage goes (until V1 blows up from overvoltage around 1500 volts or so).

Second-hand 6L6's are cheaper than 25-watt resistors, and far cheaper than 100-watt bleeders. The wattage requirement in a bleeder comes primarily from the power thrown away by excess current at highest voltage; this hookup will let a single 6L6 bleed a 750-volt supply, without wasting any current either.

CHAPTER

RTTY HANDBOOK

8

Tuning

The tuning indicator shown in Fig. 8-1 operates directly from the receivers audio output. It can be used as is if audio is taken from the high impedance phone output of the receiver. If a low-impedance speaker output is used then an 8-ohm to 10K step-up transformer will be required to insure an adequate signal amplitude at the grid of V1.

DOUBLE-EYE TUNER

An inexpensive two-shadow tuning eye tube is used. One plate tunes in mark for RTTY, or white in the case of FAX. The other plate tunes in space for RTTY, or black for the other mode. Two sets of filters are required, using either 88-mh toroids or 100-mh television-type inductors. The FAX filters are resonant at 2300 Hz (white) and 1500 Hz (black) respectively. RTTY filters are tuned to 2125 and 2975 Hz. An additional 2295 Hz filter is included for tuning in narrow-shift RTTY.

Filter tuning is most critical of course and should be within a few hertz. You can either purchase the tuned filters from hams engaged in the business or tune your own if equipment is available. If you decide to tune the filters, a frequency counter, stable audio oscillator, and high-impedance electronic voltmeter with good frequency response is required. The equipment setup is shown in Fig. 8-2. A 2-meg resistor is used between the oscillator and the tuned circuit under test to insure that a high Q is maintained.

Most audio oscillators have a fairly low-impedance output (on the order of 500 ohms) which would decrease the Q of the tuned circuit. As a result, the tuning would be broad. The capacitor should be a high-grade mylar type. Tolerance of the capacitor is not critical since tuning will depend on adding or subtracting turns from the inductor or varying the tuning slug if a 200-mh TV width control is used.

Fig. 8-3 shows capacitor values for 88- and 200-mh inductors. Set the audio oscillator to the desired filter frequency

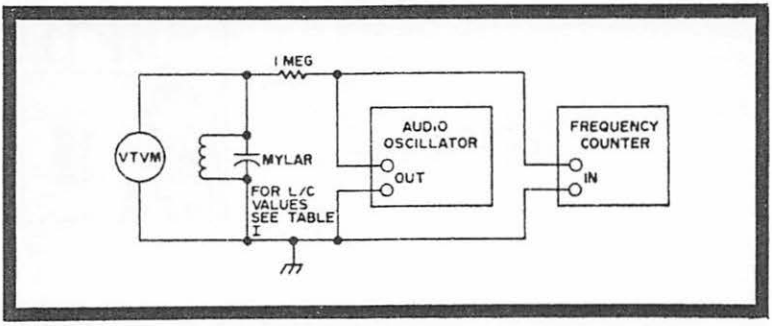


Fig. 8-1. Circuit and equipment setup for tone calibration.

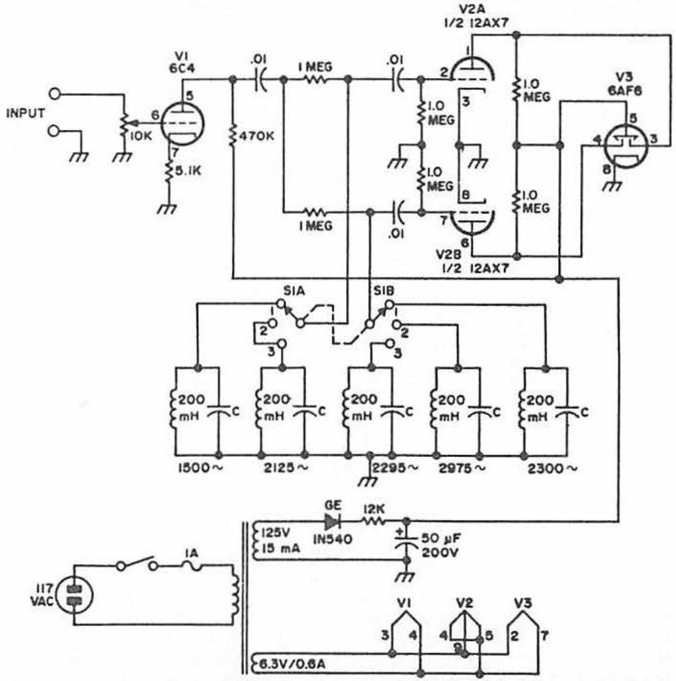
with the aid of the frequency counter. Vary the oscillator amplitude until the voltmeter (AC scale) indicates an arbitrary voltage. A low-amplitude output from the oscillator is preferred to minimize the possibility of saturating the inductor. Tune the filter for a maximum indication on the voltmeter. This insures parallel resonance.

Because of the high Q of 88-mh toroids, it might be necessary to swamp it with a low-value resistor, allowing slightly broader tuning. TV width coils, if used, are broad enough, making the resistor unnecessary. However, if 88-mh toroids are used, place a 150-ohm resistor in a series with one leg of each inductor. If the filters are too sharp, stations with shifts slightly divorced from the standard will be out of the bandpass and not received. The described technique can also be used for tuning RTTY demodulator filters.

When an RTTY station is received and the receiver is tuned to mark (2975 Hz), one of the tuning-eye shadows will close. A space signal will close the other shadow. The white or black transmission from FAX (2300 and 1500 Hz) will close its respective shadow. Shadow width is controlled by the 10K potentiometer which varies the grid drive at V1.

ADAPTER TO SIMPLIFY TUNING

To get the best possible operation from RTTY gear, it is important that the receiver be correctly tuned to the incoming signal and that the shift on the FSK be correct. Fig. 8-4 illustrates graphically the proper relations between the receiver tuning and bfo setting that we need for best results. In



S1-D.p.t.t.	
SWITCH POSITION	MODE
1	FAX
2	WIDE SHIFT
3	NARROW SHIFT

Fig. 8-2. Double-eye tuning indicator.

Frequency, Hz	L, mH	C, μF
1500	88	.125
2125	88	.068
2295	88	.047
2300	88	.047
2975	88	.033
1500	200	.047
2125	200	.022
2295	200	.022
2300	200	.022
2975	200	.015

Fig. 8-3. Capacitor values for 88- and 200-mh inductors.

the correctly tuned illustration, an optimum IF selectivity curve is shown, with a bandwidth just wide enough to pass the entire FSK spectrum. Obviously, the tuning is most critical for this bandwidth. Note that the bfo should be set 2550-Hz away from the center of the IF passband. It doesn't make any different whether the bfo is above or below the IF, this will only swap the mark and space tones. The incorrectly tuned example shows the bfo set too close to the IF. Here, the space frequency will fall off the edge of the IF curve causing it to be weaker than the mark signal. This will result in errors, particularly when noise or fading is present. If your receiver has a wider passband 1200 Hz, the tuning will not be as critical. However, it is very desirable to have the FSK signal centered in the passband. We then have the maximum tolerance for drifts, mistuning, etc.

So we see that we need to do two things. First, get our bfo settings correct and, second to be able to accurately tune in the RTTY signal so that it is in just the right place in the IF passband. The answer to both these requirements is a good tuning indicator. An additional advantage of a good indicator is that we can set the shift on our transmitter correctly.

Types of Indicators

Many different devices have been used for RTTY tuning indicators. A list might include:

1. Zero-center meter across discriminator load
2. Neon bulbs on keyer output
3. Electron-eye tubes on the mark-space detector output

4. Variable-angle scope display
5. Flipping-line scope display.
6. Detected-pulse scope display
7. Scope cross-pattern

Most RTTY'ers have their favorite indicator. However, the display which seems the easiest and fastest for the newcomer to learn to use is the cross-pattern oscilloscope. This method was originated by Merrill Swan, one of the pioneers in ham RTTY. In this type of indicator, a correctly tuned signal with correct shift will produce a perfect cross on the scope face. As you tune through an RTTY signal with the receiver dial, the cross will first be small. It will attain its maximum size when the signal is correctly tuned; and, then, will get small again as we detune. If the frequency shift of the

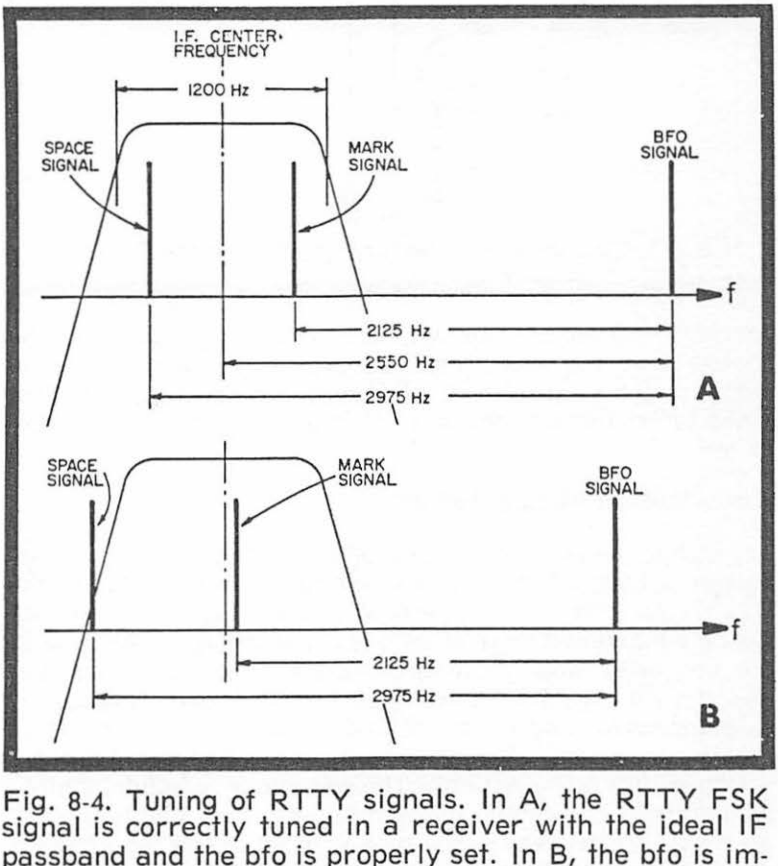


Fig. 8-4. Tuning of RTTY signals. In A, the RTTY FSK signal is correctly tuned in a receiver with the ideal IF passband and the bfo is properly set. In B, the bfo is improperly set with the mark signal in the IF passband and the space signal out of it.

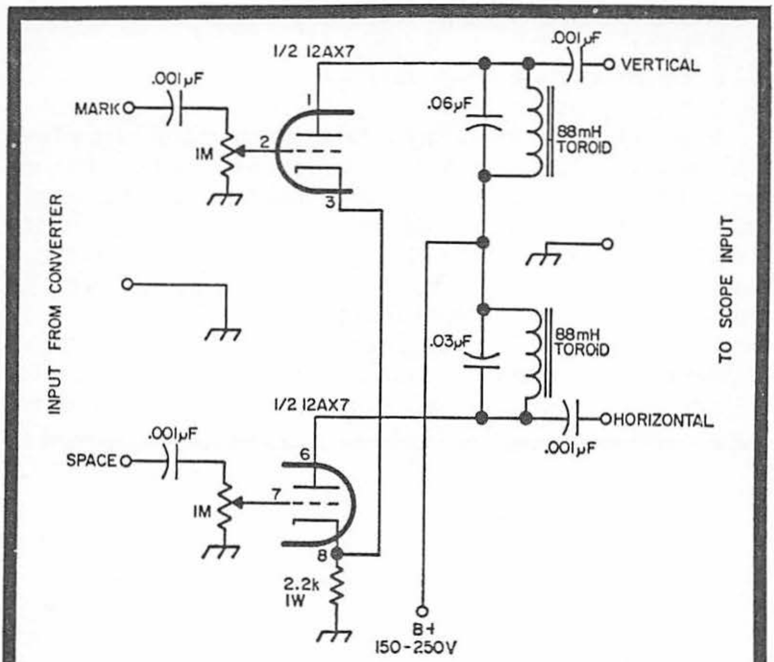


Fig. 8-5. Circuit of a scope tuning adapter for RTTY.

received station is incorrect, one arm of the cross will be shorter than the other. You can experiment with your converter to determine if a signal with the wrong shift prints best when tuned for maximum mark maximum space or in between.

Cross-Pattern Tuning Adapter

Either a standard oscilloscope can be used as a cross-pattern tuning indicator or a scope tube can be added to the RTTY converter. The simplest way of driving the scope is to feed the horizontal input from one set of the tuned circuits in the converter and the vertical input from the other set. However, most RTTY converters have relatively low-Q circuits for separating the mark and space tones. The result is that we get crossed ellipses instead of crossed lines. Accurate tuning is thus more difficult. The solution is to build a simple adapter whose circuit is shown in Fig. 8-5. This adapter can be used with almost any audio-type converter and can drive a standard oscilloscope. It has sufficient gain to drive the deflection plates of a 2- or 3-inch cathode ray tube directly. The

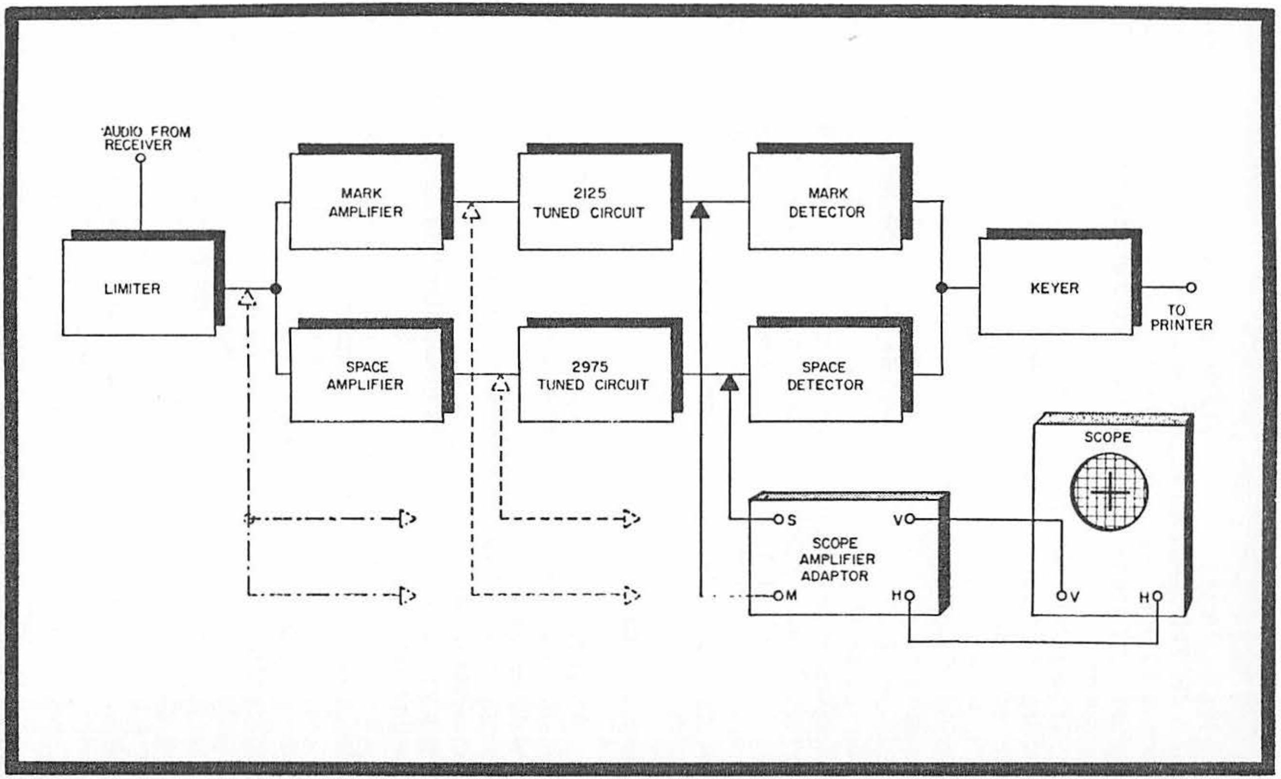


Fig. 8-6. Possible scope adapter connections for a typical audio-type RTTY converter.

adapter is very simple, consisting of two highly-selective circuits and a dual triode amplifier.

High-Q toroids are tuned to the two audio tones, which are normally 2125 and 2975 Hz. These are driven by the two sections of a 12AX7, which has a high plate resistance. The grids of the 12AX7 are fed from the two tuned circuits in the RTTY converter.

It is rather difficult to give exact instructions on connecting the tuning adapter to the many different types of converter circuits in use. The best procedure is to experiment with your particular unit until you find the points which produce the cleanest pattern on the scope indicator. However, we can give some general suggestions. Fig. 8-6 shows a block diagram of a typical audio converter consisting of a limiter followed by the audio tone filters. Two amplifiers feed the detectors whose outputs are combined to drive the keyer circuit. Points which may be suitable for connecting the adapter are indicated. Note that the mark and space adapter inputs are tied together when connecting to the limiter output but are separated when connected beyond the converter filters.

Scope Indicator

A conventional oscilloscope can be used in conjunction with the cross-pattern adapter as the indicator. However, for a few dollars, a separate scope indicator can easily be built. Fig. 8-7 is a basic circuit which uses the power supply in the RTTY converter or other existing supply for the scope high voltage. The scope tube can be a 2AP1 or similar type, available from surplus for two or three dollars. A separate filament transformer should be used since the cathode is several hundred volts above ground. The anode voltage is obtained by deriving a negative voltage from one side of the high voltage transformer in the converter power supply. This voltage will be approximately equal to the peak AC and may give sufficient brightness depending on the particular scope tube and the transformer in your power supply. If the pattern is not bright enough, disconnect the ground from the voltage divider network and return to the B+ voltage from the power supply, as shown by the dotted lines. This will put the negative supply in series with the positive supply. Be careful not to ground the various pots, and use insulated shafts for safety.

No centering controls are shown, since most scope tubes will have adequate deflection plate alignment for this use. However, if the pattern is too much off-center, it can be corrected by use of a small permanent magnet. The magnet is

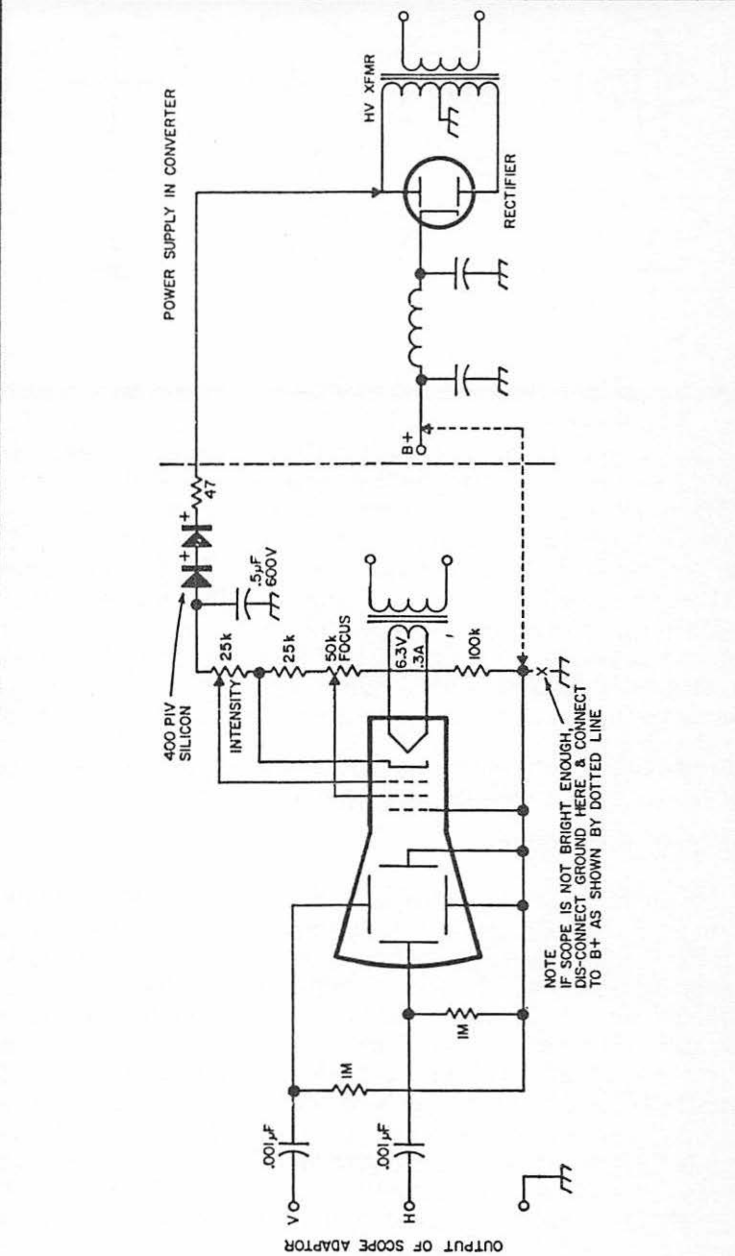


Fig. 8-7. Simple scope indicator for RTTY tuning unit.

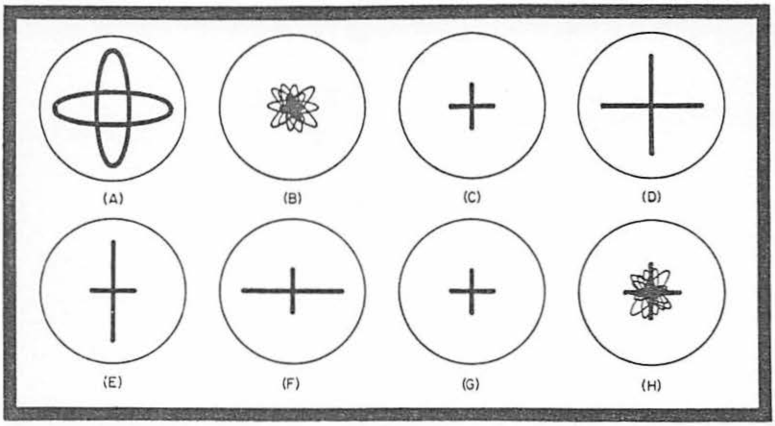


Fig. 8-8. Examples of RTTY tuning scope patterns for various signal conditions. All patterns except A are obtained using the scope tuning adapter. A Ellipse cross pattern from converter using low-Q tuned circuits. B, Pattern with no FSK signal, noise and speech only. C. Correct shift FSK, improperly tuned. D. Correct shift FSK, correctly tuned. E. FSK signal with incorrect shift, peaked on mark. F. FSK signal with incorrect shift, peaked on space. G. Narrow shift FSK correctly tuned. H. Weak FSK signal with noise (signal not limiting) correctly tuned. The text discusses the easiest method.

moved around near the scope tube and taped to the chassis or panel at a spot which gives proper centering.

Aligning the Adapter

With the adapter connected to the scope indicator, the two toroids in the adapter can be trimmed to exact frequency. Probably the easiest and cheapest toroids available are the 88-mh loading coils which can be bought for less than $1.00 each, although any toroid from 50 to several hundred millihenries is suitable. The approximate values of capacitance required for 88-mh are shown. With an accurate source of audio tuned to 2125 Hz and fed to the grid of V1A the capacitance is trimmed across L1 to obtain the longest possible line on the scope. Similarly, 2975 Hz is fed to the grid of V1B and the capacitance across L2 is varied to produce the longest line. Alternately, the capacitances could be fixed and turns removed from the toroids.

After the adapter is properly tuned, it can be connected to the RTTY converter. The scope gain controls are then ad-

justed for the desired size cross-pattern and the unit is ready to use. Tuning in an RTTY station is extremely simple. Just tune for maximum cross size. Fig. 8-8 shows the patterns obtained for various conditions.

Receiver Tuning Hints

We mentioned earlier that it is important to have the bfo correctly set. We would like to get our receiver set up so that the bfo is in the proper relationship to the selectivity curve of the IF as illustrated in Fig. 6-4. Once this is done, we should then tune in RTTY signals using the main tuning dial, leaving the bfo fixed. The following procedure is suggested to get your receiver set in this manner. Once this is done, tuning RTTY signals with the tuning indicator is easier and faster than CW or SSB.

1. Turn on your vfo to provide a steady carrier.
2. With AGC on and bfo off, tune in your vfo signal and carefully peak the signal with the S-meter. You now have the vfo signal centered in your IF passband.
3. Turn on the bfo, and zero-beat the vfo signal with the bfo pitch control. The bfo is now tuned to the center of the IF passband.
4. Detune your bfo 425 Hz higher in frequency. This will produce a 425-Hz beat note with the bfo. You can set this ac-

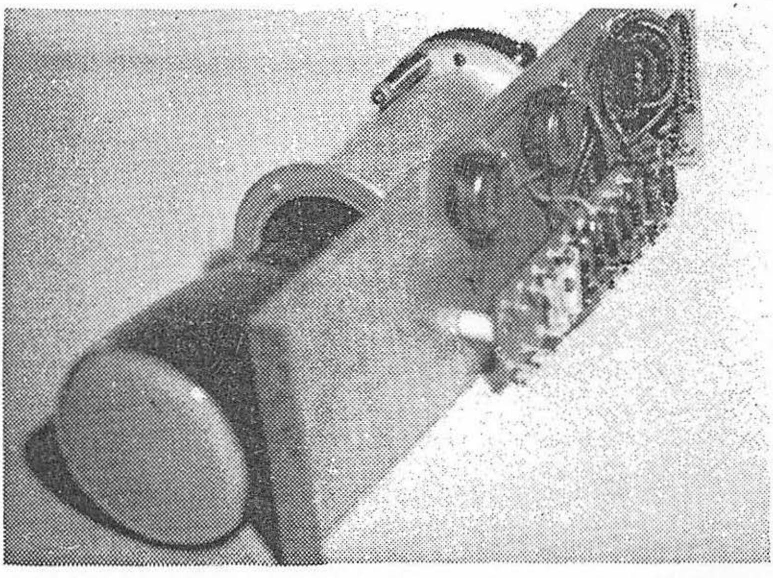


Fig. 8-9. Scope monitor.

curately by operating your FSK (switching between mark and space) and carefully tuning the vfo so you get the same 425-Hz tone from mark condition as from space condition. This is true since the bfo is in the center, while the space frequency is 425 Hz lower and the mark frequency is 425 Hz higher.

5. Retune the bfo pitch control to produce a 2125-Hz beat against one frequency and 2975-Hz beat against the other frequency. This setting can be made most accurately by operating your FSK and adjusting the bfo pitch control for the maximum size cross on the tuning indicator scope. Some bfo controls may not have quite enough range to get the 2975 Hz beat, but this can usually be corrected by a slight adjustment of the trimmer on the bfo coil.

6. When the proper bfo setting is found, make a mark on the receiver panel of some kind so that you can always set the bfo pitch control to this position. Always tune in FSK signals with the main tuning dial and not with the bfo control.

Setting Transmitter Shift

By using a cross-patterns tuning adapter, transmitter frequency shift can quickly be set to the correct value. With the transmitter exciter on and in the mark condition (keyboard closed), tune in the signal with the receiver until the scope line representing mark has its maximum length. Then simply depress the break key on the teletype machine, and adjust the shift control to obtain the maximum length line on the scope representing space. Also, zero-beating another RTTY station is very fast and easy with the tuning indicator. When calling another station, you zero-beat his mark signal by tuning your vfo in the spotting mode to produce the same line on the scope. You can also operate your break key for a quick check on your shift.

A MONITOR SCOPE

Tuning RTTY signals without the use of a scope monitor is a difficult job. So, in designing the complete converter it is well to include a scope (Fig. 8-9).

In order to make the circuit simple, no amplifiers are used. There is sufficient signal delivered to the deflection plates of the 3BP1 for good scope presentation.

In Fig. 8-10, the power supply connections are shown, since the power supply provided for the converter circuit is ample to handle the scope monitor. Construction details can be seen in the accompanying photo. All wiring is done on one side

of the sub-chassis by use of a terminal board. The two pots shown are the horizontal and vertical controls. The focus and intensity controls are mounted on the front panel.

The sub-chassis is 3¾ inches wide by 12¾ inches long, with a one half inch flange at each end.

After completing the wiring as shown, fit the sub-chassis into the overall chassis; and fix into position with self-tapping screws. This part of the converter is dressed up by using a Millen bezel No. 80073.

After wiring is completed, turn the converter power supply on and adjust the vertical and horizontal controls. With an incoming signal applied, adjust the focus and intensity controls.

The pattern which should appear when the signal is properly tuned is a cross.

Tuning by the scope monitor is easy, as you have a visible indication of when proper tuning is achieved. The scope presentation will be a perfect cross, if the shift of the transmitting station is correct. Practice will soon give you the necessary skill to recognize the proper pattern when it appears on the scope.

A SIMPLE SCOPE

Tuning in an RTTY signal is virtually impossible unless some form of tuning indicator is used. The most common

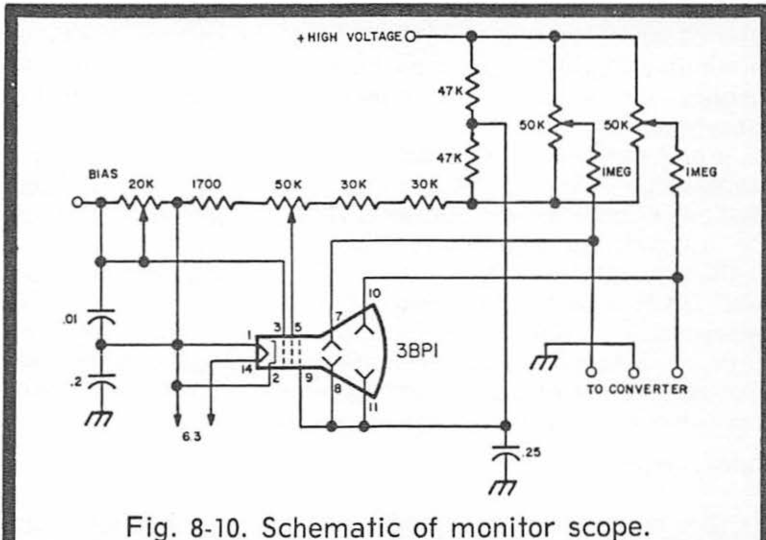


Fig. 8-10. Schematic of monitor scope.

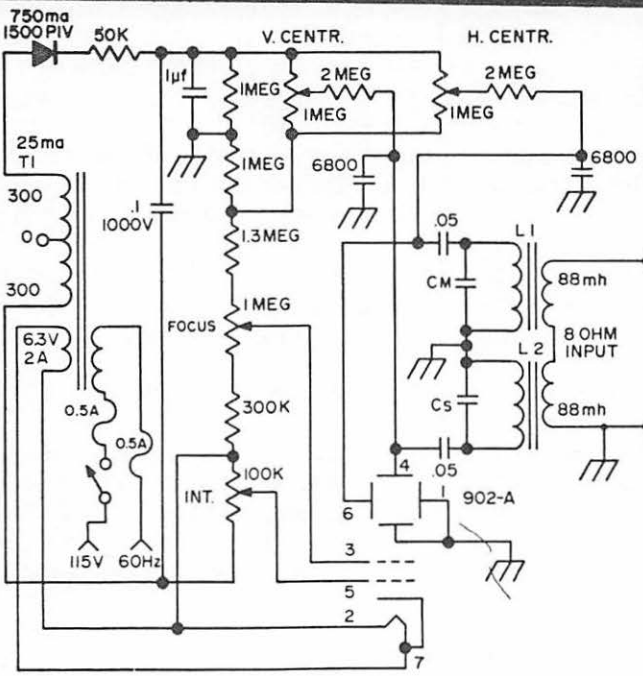


Fig. 8-11. Schematic of simple scope.

indicators are the tuning eye tube, meter, and cathode ray tube. Of these, the most flexible by far is the cathode ray tube. In addition to providing tuning information, it can be used to identify interference and check on proper transmitter operation.

For some reason, most amateurs shy away from building oscilloscope indicators. Actually, they are no more difficult than any other electronic project. This little indicator is about the ultimate in simplicity. This is due to the current availability on the surplus market of the 902A cathode ray tube. It has a deflection sensitivity of 90 volts per inch. It is thus possible to obtain adequate deflection without amplifier stages. In addition, its high voltage requirements are modest, allowing the use of a small cheap power transformer (300-0-300v) in a half-wave rectifier configuration.

Construction

The scope is constructed on a 6" x 9" x 2" aluminum chassis. The circuit (Fig. 8-11) is divided into two sections. The

power and oscilloscope control components are mounted on the panel and chassis. The input transformer and tuned filter networks are constructed on a 2 x 5 inch piece of vector board which is mounted underneath at the rear of the chassis on stand-offs. Leads to this board should be long enough to allow the board to be slid out of the chassis for tuning.

If, due to the small size of the cabinet, it is necessary to mount the transformer with the core parallel to the axis of the cathode ray tube, any unwanted deflection can be cured by a tube shield made from a 4-inch length of 2-inch diameter galvanized water pipe. After cutting, the shield is de-burred and painted black (Fig. 8-12).

Typical receiver output lines have an 8-ohm impedance whether they are from the HF communications receivers, or the VHF FM system. A standard 88-mh toroid is used for a combination tone filter and step-up transformer. To match the 8-ohm line a primary winding of 35 turns of No. 22 enameled wire is wound over the existing turns of the toroid.

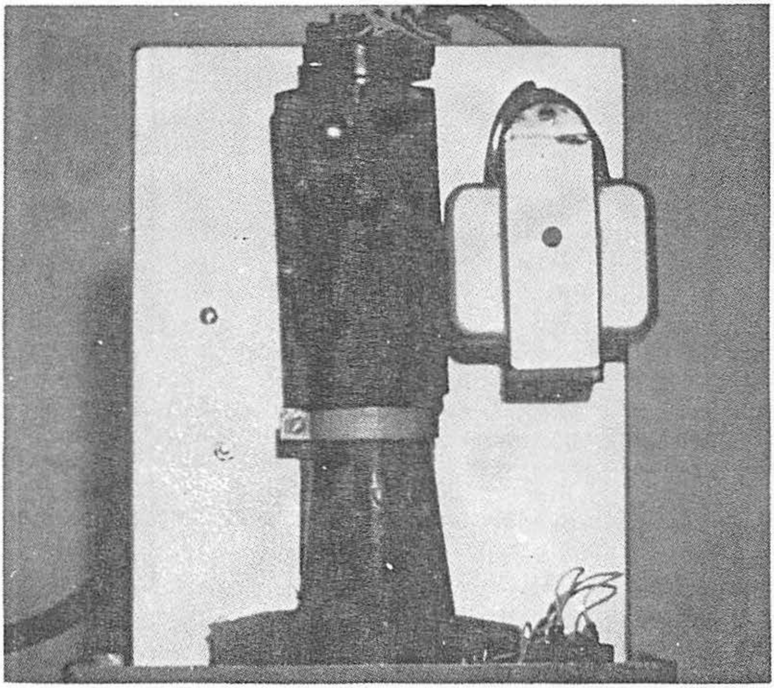


Fig. 8-12. Top view showing water pipe shield installed. Note the small foam rubber cushion at the front of the tube.

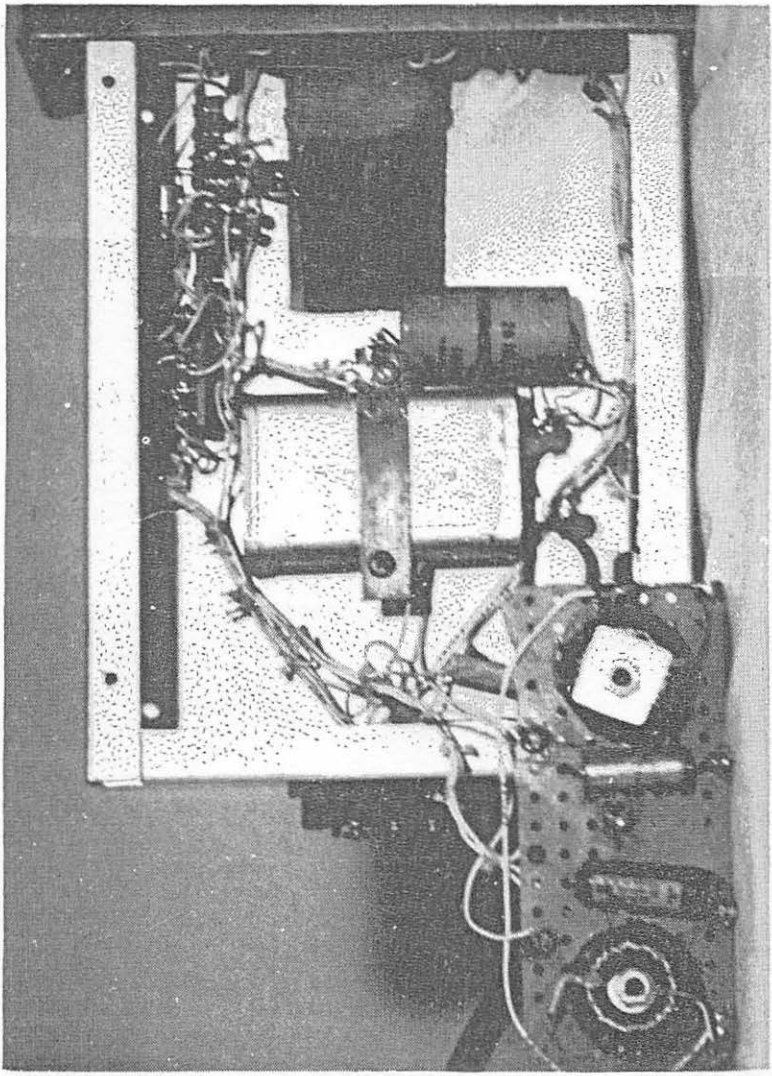


Fig. 8-13. Bottom view showing filter board removed for tuning.

Testing Scope Operation

After checking the wiring, turn the scope on and allow it to warm up. Advance the brilliance control until a spot appears. Then, sharpen the spot with the focus control. Center the spot with the centering control. Connect an audio signal generator

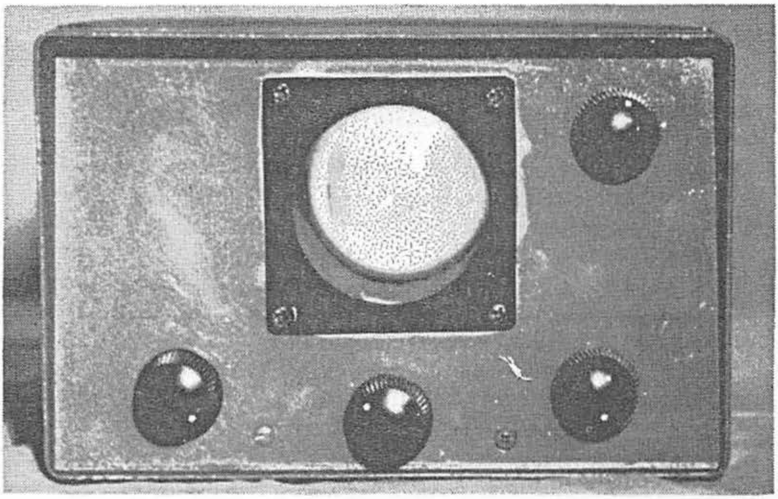


Fig. 8-14. Front view of simple scope.

to the input. Remember, most audio oscillators have a 600-ohm output impedance, so a matching transformer should be used.

There are several ways to tune the filters (Fig. 8-13). Use of a counter to check the frequency of your audio oscillator is well worth the trouble, if you can obtain this instrument.

The easiest way to tune the filter is to use a value of 0.033 mfd for the space capacitor. With your oscillator set at 2975 Hz, tune the circuit to resonance by removing turns from the toroid.

About 4 turns per Hz is a good rule of thumb. Next tune the mark filter using a 0.066 mfd capacitor to start with, and the same procedure. The audio oscillator should be set to 2125 Hz.

An alternate method is to substitute capacitors until the desired frequency is reached. This is simpler than removing turns only if you have a large selection of capacitors. A good grade of mylar or paper capacitor should be used.

If you cannot get your hands on audio oscillator, the output from a friends AFSK oscillator can be used.

In operation, the input line is bridged across the 8-ohm input to the TU. The receiver bfo is turned on and the desired RTTY signal tuned in until a distinct cross is obtained on the CRT (Fig. 6-14). The audio gain is adjusted to obtain the desired height.

CHAPTER

Reading and Care of Tape

Have you ever read the tape that you just printed off your Model 19? Here is an interesting way of reading it, using the binary system. Look at the tape from the top with two holes to the left of the sprocket and three holes to the right of the sprocket. Group the two to the left, to read 2-1 and the three to the right to read 4-2-1.

TAPE READING IN BINARY

Now all you have to do is add up all the holes that are punched to the left as one digit and all the holes punched to the right as one digit. This will give you two digits which can be converted to a function on the machine.

Example 1. If all holes are punched, add up the two weights on the left (2+1 equals 3). Now add up all the weights on the right (4+2+1 equals 7). You now have two digits which put together is 37. Look on Fig. 9-2 and you find 37 would be the LTRS function.

Example 2. No. 1 hole on the left with a weight of 1, and the No. 2 hole on the right with a weight of 2. Put together, this is 12. Look on Fig. 9-2 and you find 12 is the function R.

Example 3. No. 2 hole on the left weight of 2 and the number 4 and 1 hole on the right summed is a weight of 5. Putting the two digits together gives 25. Look on Fig. 9-2 and find 25 is the Y function.

If you study the table you can see that there are no wasted functions. There is a possible combination of 32 functions. If you add the 26 upper case operations we have 58 different possibilities. When reading the tape you should know if the tape is in upper or lower case. This should not create any problems. You could add one more function which is the blank.

CARE OF TAPE

For those who want to keep, store and retrieve tape, there seem to be several problems that are encountered. A few tips may be in order.

There are several types of commercial winders available. If you use a lot of tape, it is convenient to use these, preferably one that is motor-driven. One can be used for winding from the TD or reperf and the other for the necessary unwinding. On a long tape, put the loose tape from the unwinder in a large wastebasket below the TD and then forget it, as it comes out fine. Chadless tapes may catch on themselves and require a bit of watching. If you do not have a winder, you can learn to wind the tape in a figure eight about your thumb and little finger with the start at the beginning of the bundle. In this way, the tape will not be twisted and will pull off the tape bundle with but a little help from you. To keep the ends free and windable, pull about a foot of blank tape from the reperf (Fig. 9-3) at both ends. If you are thinking of building a winder, try to keep the center hub at least three inches in diameter as too tightly wound tape resets the chads in the chadless type back where they were and will not permit it to run properly in the TD. If you use the figure eight wind, tuck the free end into the bundle loop. With rolls from the winders, use a small bit of solder to hold the roll together. If you have trouble learning to figure eight wind, get a local commercial Teletype operator to show you the simple technique.

Storage and retrieval are the real problems. One approach is to separate the tapes into subjects and store them in large plastic bags with ten or fifteen tapes to a bag. This keeps the tapes from drying out and lets you find any one tape without too much time. The title of the material on the tape,

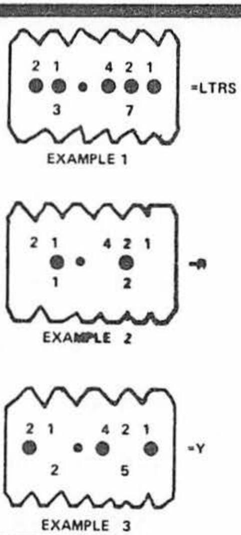


Fig. 9-1. Tape punches.

01 =	T	20 =	E
02 =	CAR RET	21 =	Z
03 =	O	22 =	D
04 =	SPACE	23 =	B
05 =	H	24 =	S
06 =	N	25 =	Y
07 =	M	26 =	F
10 =	LINE FEED	27 =	X
11 =	L	30 =	A
12 =	R	31 =	W
13 =	G	32 =	J
14 =	I	33 =	FIGS
15 =	P	34 =	U
16 =	C	35 =	Q
17 =	V	36 =	K
00 =	BLANK	37 =	LTRS

Fig. 9-2. Binary conversion table.

marked on the lead end with a heavy marking pen, also helps to speed identification. Making and keeping a list of the tapes in the same order as stored also helps and can be used to tell the others what you may have on hand that may be of interest to them. The marking pens will also write on the plastic bags that can then be stored in a drawer or cabinet.

Tape Repairing

Sometimes when making a tape or reperfing, your supply roll will run out; or you may have a break or tear in a tape. The simple side tear problems are easily fixed with a bit of Scotch tape trimmed to the edge of the tape. Otherwise, you may need a splice or patch. These are not the best solutions, and you should make a copy tape with the repaired portion, but a fairly good splice may be made with white glue to add a fresh roll to one that is running out. Overlap about an inch or so. Be sure the glue is dry. You can also punch a short piece of tape with the same characters and splice a bad spot in the middle of the tape. With the chadless types, they may be merely run together in the TD to hold the splice. But if you want to keep the tape for any time, take the time to make a new one.

When running a long tape on the air, do not forget the need to ID at least every ten minutes. If you have the narrow shift CW ID facility, that is fine and will lock up most of the machines. In the absence of the narrow shift (about 100 Hz), stop the TD and let the steady mark tone stay for a few

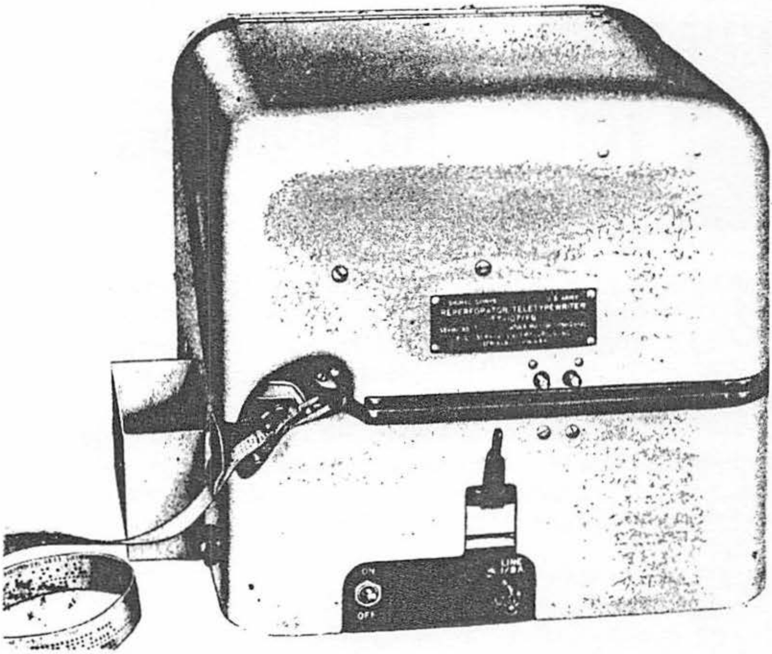


Fig. 9-3. TT-107-FG Kleinschmidt reperforator.

seconds, then CW ID and repeat the steady tone to permit the other fellow to hit his standby switch and not interrupt his print. When making tapes, make them as short as possible, and leave out all those lines across the paper that are in some tapes, such as "CW ID follows" and similar unnecessary characters or language.

CHAPTER

RTTY HANDBOOK

10

FCC Regulations

The FCC permits radio teletypewriter operation in all but the 160-meter band. On the lower frequency bands, operations are limited to either make-break or frequency shift keying (FSK). Make-break transmission never achieved much popularity. In this system, the transmitter is keyed on and off (CW) with either the mark or space signal. Many tests were conducted with this system of transmission before FSK was permitted on the lower frequencies, but these tests always did more to discourage this type of transmission than to encourage it.

AVAILABLE BANDS

On the 80-meter amateur band operations are permitted for send FSK (F1) emissions from 3500-3800 kHz. In practice, RTTY signals are rarely transmitted more than a few kHz from the net frequency of 3620 kHz. This channel was originally chosen when the band was opened to RTTY because it was the only channel that did not have a large number of traffic handling networks on it. Considerable interference coming from South American commercial stations on this channel had made it unusable for regular CW nets, so RTTY moved in and made use of it. It was felt that RTTY would interfere with CW ragchewing if used in the 3500-3600 kHz segment. The 3600-3700 kHz segment was completely filled with traffic nets except for 3620. Above 3700 were Novice and Canadian phone.

On 40 meters there was more room for choice. The original selection, out of the allocation of 7000-7200 by the FCC, was 7140 kHz. The reasoning was that RTTY should keep away from the heavy ragchewing and DX concentration of CW stations down at the low end. Above 7150 was Novice. So 7140 was selected. Radio Moscow has not contributed to the popularity of this choice in recent years, and there is a strong movement to change it to 7040 kHz.

20 meters is the most important DX band., The FCC allocated us from 14,000 to 14,200 kHz for F1. Activity has

centered around 14,090. On most nights you can hear some interesting DX coming through on RTTY...a ZK1 for instance.

15 meters, when open, is a fine RTTY DX'ing band. The FCC has allocated 21,000 to 21,250 kHz for F1, but activity will usually be found within a few kHz of 21,090.

10 meters is a different story. Since it will be some years before this band will be any practical use for long distance communications, it probably will make no different to the average operator that normal CW allocations are now available for FSK RTTY on ten meters. FSK is permitted from 29.0 to 29.7 MHz. The lack of activity on this band has left no recommended channels, though 29.090 MHz would be consistent with RTTY channels on 15 and 20 meters.

On 6 meters RTTY can use either FSK or AFSK from 50.1 to 54.1 MHz. This is the lowest frequency band for use of AFSK. There is, at this time, relatively little RTTY operation on 6 meters.

The two-meter band from 144.0 to 147.9 MHz is available for both FSK and AFSK operation. While there is little activity in this band on RTTY, you may hear it on the MARS channels just below the band. AFSK is generally used on two meters.

F-1 LIMITATIONS

The FCC stipulates the character of the RTTY emission rather carefully. It states that, "A single channel five-unit (start-stop) teleprinter code shall be used which shall correspond to the International Telegraphic Alphabet No. 2 with respect to all letters and numerals (including the slant sign or fraction bar); but special signals may be employed for the remote control of receiving printers, or for both purposes, in "figures" positions not utilized for numerals. In general the code shall conform as nearly as possible to the teleprinter code or codes in common commercial usage in the United States."

The FCC has recently passed an amendment to its rules to allow hams to use speeds of 60, 67, 75, and 100 words per minute on teletype equipment. Previously, amateurs were limited to 60 words per minute. This limit was mainly imposed by the type of equipment available. Now, however, more surplus TT gear is being released from commercial service; and increased speed is now possible. Arguments in favor of the proposal included greater message handling capability and more efficient use of airtime.

Another rule requires the signing of the call letters of the sending and receiving stations at the beginning and end of each transmission, unless operating break-in (transmissions

less than three minutes each), and at least once every ten minutes.

NARROW SHIFT

The original FCC regulations for RTTY did not permit anything but 850-Hz shift. This was a little restrictive to the experimenters, and eventually the regulations were changed to permit any shift up to 900 Hz. Many different frequency shifts have been tried, with 160 and 170 Hz turning out to be the most popular. The 160 Hz shift is easy to determine since WWV transmits 440 and 600 Hz which are 160 Hz apart, providing a fine standard for reference. Commercial narrow shift uses 170 Hz, so this too has been used on the amateur bands. Narrow shift is seldom used. The stations on the amateur frequencies will be using 850 cycles.

CHAPTER 11

RTTY Art

The human voice had been traveling over wires for nearly a half century before a method was devised to transmit photographs or visual information. Many attempts were made to achieve this goal but will little success until the development of a satisfactory photoelectric cell in 1921. While the photocell was being studied as an instrument to analyze a photograph, Mr. E. F. Watson, an engineer at the A. T. & T. Co., suggested that a satisfactory representation of the different shades of a photograph could be obtained by using a typebar printing telegraph machine with typebars fitted with different sized dots or even with some of the regular printer characters.

EARLY EXPERIMENTS

In order to test whether the representation of a picture in this manner would be satisfactory, a photograph of a keyboard perforator was analyzed point by point and the shade of each point estimated. A reproduction of the original photograph was made by reproducing these shades in dots of different sizes on a separate sheet of cardboard. This was all done manually, and thousands of unit areas were examined and reproduced in this way to make the complete picture. The result was considered excellent. A typewheel printer was modified to carry a photocell analyzing device which could read the density of a photograph on a transparent film. The density readings were converted to codes which were automatically punched into a paper tape. Only five densities were assumed. This tape was then read out and printed on a slightly modified page printer (line spacing was reduced) using the letter M for extremely dark, H for dark gray, T for gray, period for light gray and space or no printing for the lightest shade.

One of the first pictures used in this apparatus was a photograph of Mr. Harry B. Thayer, president of A. T. & T. (see Fig. 11-1). The very satisfactory result was obtained on

Fig. 11-1. This picture was printed from an automatically perforated tape on April 1, 1923. (American Telephone and Telegraph Company, Department of Development and Research)

April 1, 1923. Soon experiments were carried out transmitting such pictures across the country. Another very successful receiving mechanism was developed using an automatically controlled artist's air brush. By May 1924 modulated carrier current methods and photographic film recording were successfully demonstrated. This marked the beginning of telephotography as we have known it for years. It is interesting to note that earlier experiments had been designated as techniques of phototelegraphy.

HOLIDAY GREETINGS

Apparently the idea of creating pictures from normal type characters caught on with the telegraph operators, and they began to create some of their own simple pictures manually. Early pictures were generally made up almost entirely of X's. The story goes that teleprinter operators who were assigned to duty over the holidays, particular Christmas, would cheer one another by sending their own art around the country. One of the first complex designs, a Madonna, is credited to Meyer Hill, an Associated Press operator in Baltimore. This picture was created in 1947 (see Fig. 11-2).

The tradition spread; and, with the advent of worldwide circuits, the exchange of pictures became international. Charles Reeser of AP Washington is noted for his work and was the originator of the "stained glass window" technique used in the Angel and Shepherds picture. See Fig. 11-3.

AMATEUR RTTY ART

Obviously, with the advent of amateur radioteletype, the picture transmission scheme has been carried over to this field. The first pictorial attempt by many amateurs is some form of QSL "card" transmitted right along with the QSO. These usually show the sender's call letters in block form and confirm the specific contact by manually typing in the details at a predetermined point, often spotted by a series of bells on the basic tape.

ELABORATE PRODUCTIONS

Pictures that have been flowing around the world on landline circuits gradually came into the hands of amateurs who relayed them over the air. Some of these reached people who were so impressed by them that they decided to try their own hand at keyboard composition. As on the landlines, Christmas

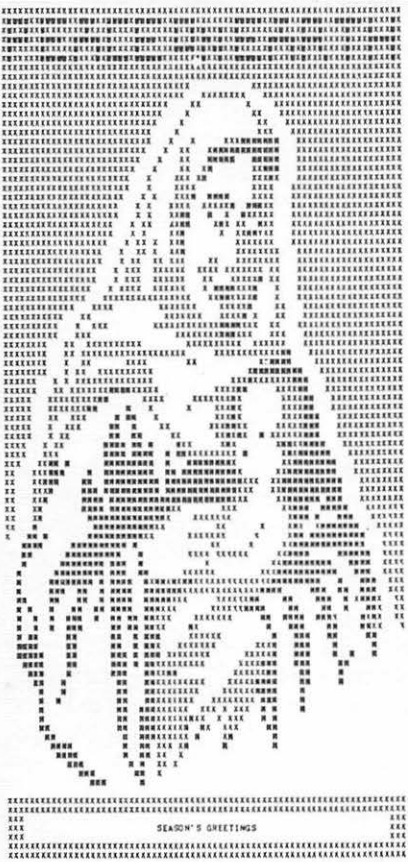


Fig. 11-2. Meyer Hill's "Madonna."

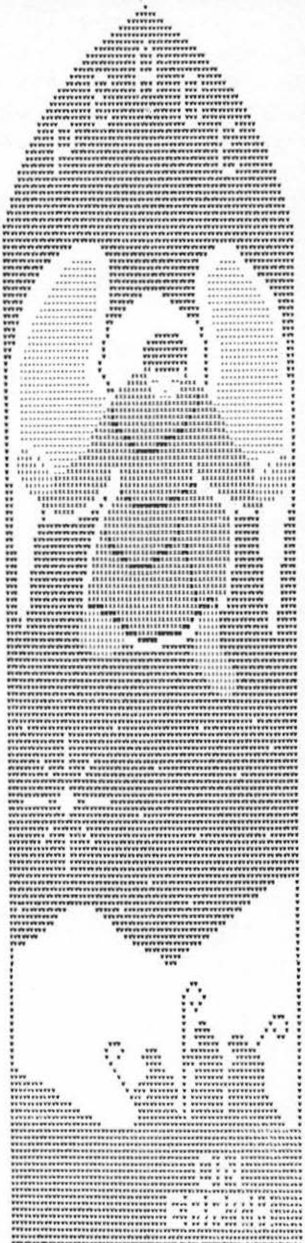


Fig. 11-3. Stained glass window technique used by Charles Reeser.

season subjects were predominant at the beginning. Gradually some "artists" became well known, each with his own particular style or field of interest. Although not a ham, Ralph Larsson of Teletype Corporation became well known for his portraits of presidents and other famous people. His pictures were relayed many times by amateurs. Probably the most famous of his series is a portrait of John F. Kennedy. (See Fig. 11-4.) Jerry Hall K1PLP originated a varied range of pictures including portarits and various holiday subjects. By this time "computer pictures" were on the scene, and Jerry adapted one of these to our 73 character per line limitation and simultaneously added some brief clothing. (Fig. 11-5) This was a very popular picture and was modified by several other people as time went on. The distaff side of the art really entered the scene with the receipt of a picture apparently originating in Sweden and known here only as "The Swedish Nude" (Fig. 11-6). This picture arrived in European format, with all of the apostrophes coming in as bells, which made a rather distorted mess of the reproduction. The corresponding tape was manually "translated" and to this time is undoubtedly the most widely distributed picture in the country.

The late Dom Rossi W3ILZ produced a very extensive collection of well known cartoon characters. Most of Dom's

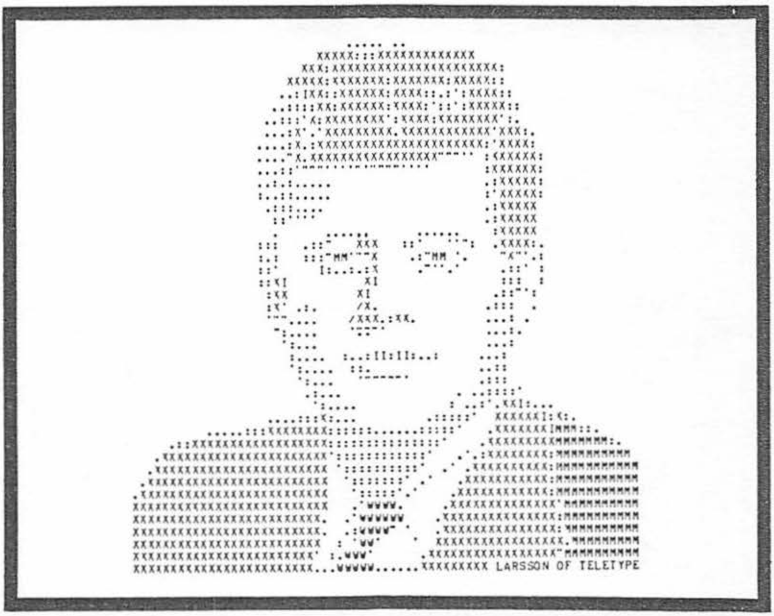


Fig. 11-4. Portrait of John F. Kennedy by Ralph Larsson.

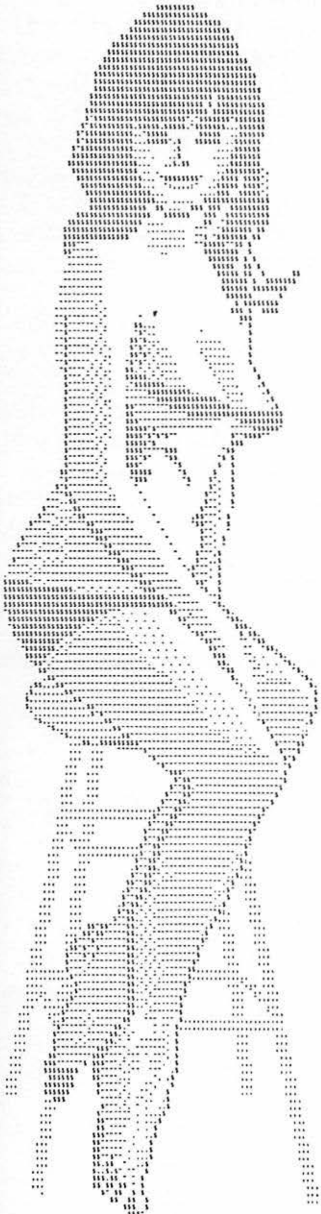


Fig. 11-6. "The Swedish Nude," probably the most widely distributed subject in the RTTY art field.

Fig. 11-5. Computer art.

work is immediately recognizable, as he soon developed a style of his own.

On the West Coast WA6TEB, W6VHF, and K6PDR were producing and relaying pictures on the VHF bands. Don Royer WA6PIR took a liking to producing RTTY art and even got his XYL, Maxine, involved as his artist (Figs. 11-7 through 11-10) Don has produced over thirty pictures, most of which are very lengthy and painstakingly shaded. He is most famous for his "gatefold" pictures which he was producing monthly for quite a while.

REPRODUCTIONS

John Greve W9DGV offered to undertake the publishing of a packet of reduced size reproductions of all the RTTY pictures that interested amateurs would be willing to contribute. Contributions of pictures trickled in; and, near the end of the year, John had enough to publish an initial packet of 50 reproductions. In recognition of the fact that some people might like to obtain tapes to run their own copies of the pictures, W9DGV also offered magnetic tape recordings which could be played through a demodulator into a printer or directly on the air on the VHF bands. John W9DGV did not have the necessary equipment to offer punched paper tape; so John Sheetz K2AGI volunteered to take on this project. Many prints and tapes were received in quite poor condition, having been relayed many times over the air and accumulating errors on the way. These tapes had to be manually checked character by character, correcting all obvious errors and removing many non-printing codes. The collection has grown to over 250 pictures at the present time, and W9DGV has produced three different volumes.

New artists are turning up regularly as interest spreads. It is difficult to estimate how many pictures must be in existence at the present time as they often tend to stay in localized areas. This is usually because the most common medium for transmission is the VHF bands. Twenty meter transmission has picked up though, and pictures are beginning to get worldwide distribution.

YOU CAN BE A RTTY ARTIST

Have you ever wished that you could make some of that RTTY art that you may have printed? It is easier than you might think.

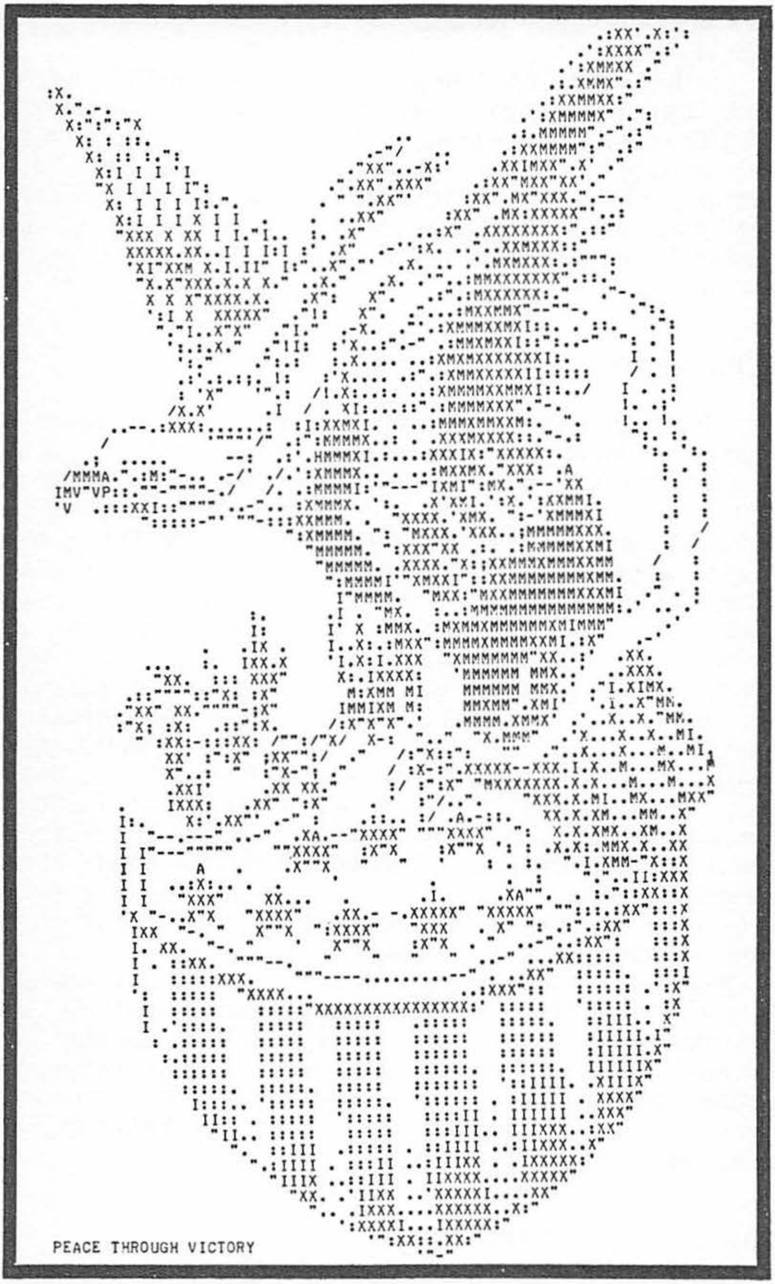


Fig. 11-7. "Peace Through Victory" originated for RTTY by Don Royer WA6PIR. Art by his wife, Maxine.

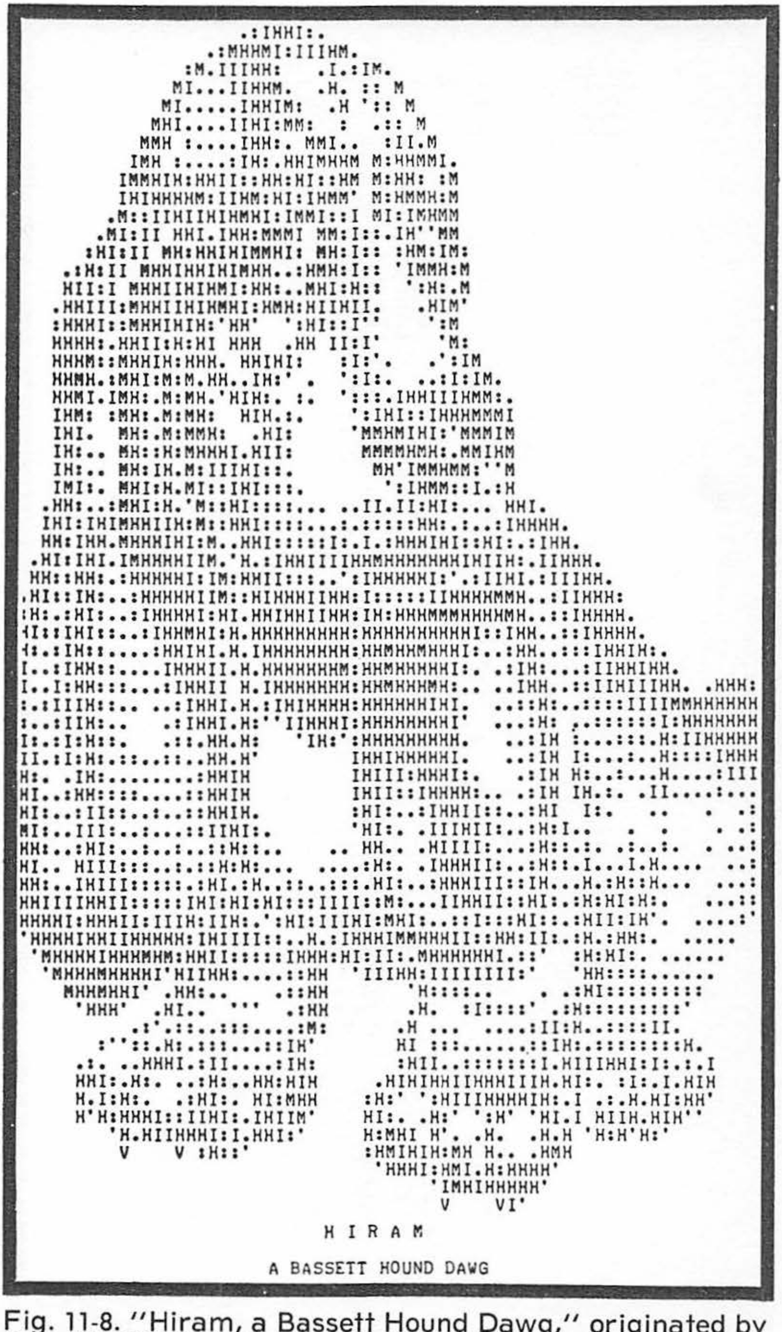


Fig. 11-8. "Hiram, a Bassett Hound Dawg," originated by Don WA6PIR. Art by XYL Maxine.

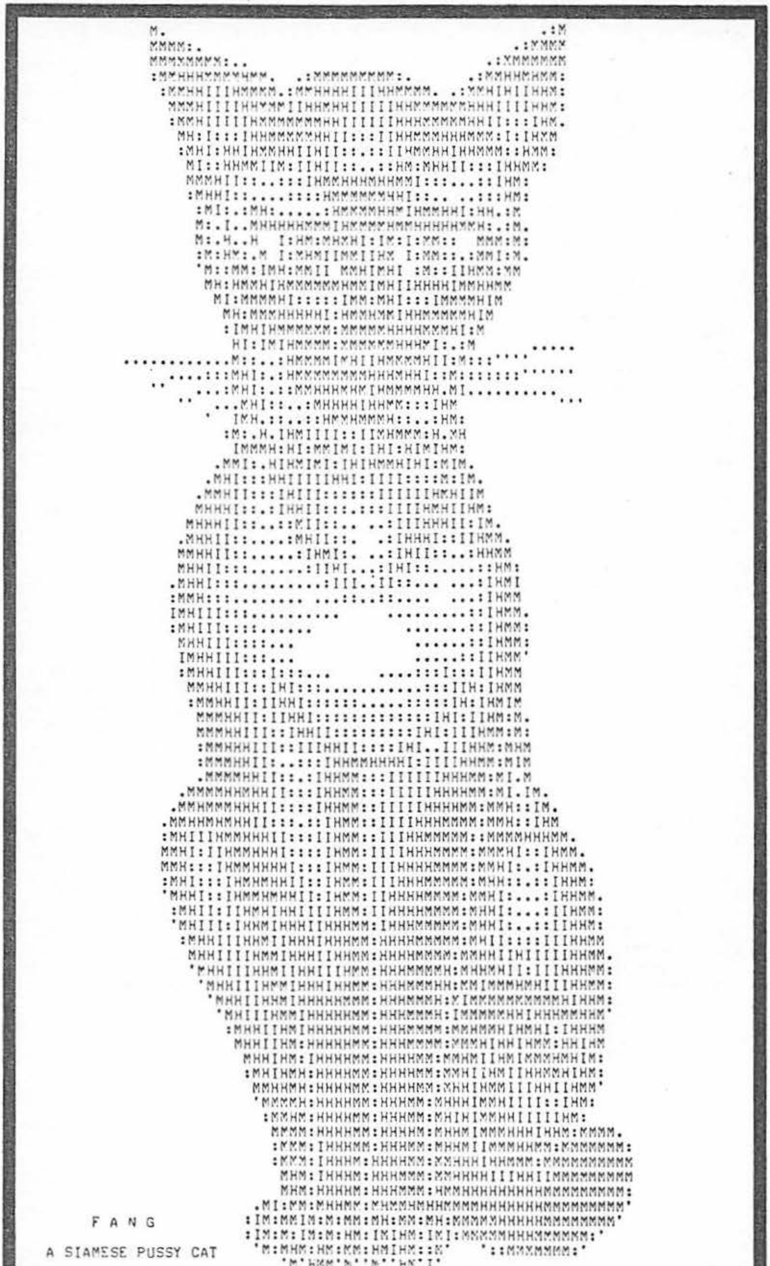


Fig. 11-9. "Fang, A Siamese Pussy Cat," originated by Don WA6PIR. Art by XYL Maxine.

Fig. 11-10. "Miss Santa," believed to be by WA9CCP. Relayed by WA6PIR.

There is much basic art work available from which RTTY pictures may be made. Cartoons, the comic strips, post cards, magazines, newspapers, centerfolds, and photographs may all serve as bases for pictures. While these may not be the right side, an inexpensive pantagraph may be used to enlarge or reduce them. A portrait of Washington was made from the etching on the dollar bill. If you have a little sketching talent, that will also help. Or enlist your wife and friends.

Prepare the Subject

Having decided on the subject and having the basic art work the right size, run about four feet of paper out of your printer. Use the center portion of the paper for your sketch or carefully tape or glue (white glue works well) the drawing or photo to the paper. Trim the edges so that all is still the same width as originally. Now take out the paper from your printer and insert the four-foot sheet with the sketch on it so that it will be presented to you as it rolls through the machine. Carefully align the edges of the paper on the platen. Use your line feed to bring the top of the sketch into view. With a little practice, you will be able to tell just where any character will strike the paper.

Overtyping

You are now ready to overtype the sketch, punching a tape as you go.

We have found that a small selection of characters is all that is really needed to produce either outlined or shaded pictures. While you may not agree with our selection, study the letters and other characters to learn their individual densities. For example, the M and W are the darkest, followed by the H or X and then by the I. Thereafter, you can use the upshifted characters such as the : or ; followed by the " or - or . and the like, depending upon where you want the print to fall. In this way, you may add the shading that you desire or leave certain areas blank like this:

MMHMHHIHII:I::.:... ...:.::I:IHIHHMHMM going from dark to light and back to dark again. Keep up this process over the entire sketch. Remove the four-foot paper with the sketch from your printer and reinsert your paper stock. Now play out the tape that you have made and see what you have. You will probably be pleasantly surprised. From this point on, take a red pen and indicate on the print where additions, corrections and any changes are to be made. Rerun the tape (having

folded the marked-up print and following it line by line, using the paper holder and line guide on the printer) and make the corrections, making a new tape at the same time. In most instances, we can now come up with a pretty good picture with a series of five or six corrected tapes.

When overtyping the sketch, it helps to have a strong light directed on the sketch in your machine. This is particularly true if the contrast of the sketch is poor, as it may be in color photos. Also, as you will not be able to see the part of the sketch below the ribbon, take a pencil and outline the areas where the shading will change from one density to another. This way you may be able to get a more complete picture the first time through. Some artists have found that it helps to make Xerox prints of the original sketches or photos and to use those for the over-typing, as they eliminate some shading and provide a black and white sketch from which to work. Keep the detail of the original art work as large as you can and don't be afraid to experiment with different letters and techniques. Clean up the tapes during the first run-through after the overtyping is finished to remove the extra characters and corrections made during the typing. To give you some idea of the time required to complete the pictures, about 20 hours are required for one that runs 30 minutes or so. Most of this time is used in making the corrections while rerunning the tape. Even after they are apparently finished, hang them across the room to see how they will look from a distance and them make the final tape with the finishing touches.

Transmitting Considerations

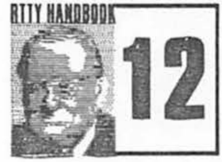
Many of the machines in use today have non-overline features so do not use overlining. Stay within a 73 character line. Start and end the tape with a series of letters, a couple of carriage returns and about ten line feeds, as this will help the receiving operator if he is making a reperf tape at his end. Also, keep in mind those who have machines that downshift on space as well as those that do not do so. If you are upshifted and then space and wish another upshifted character, put in another figures character. Of course, the same applies when you want a letter following a space after an upshifted character. At the start of each line, generally use two carriage returns, the line feed and two letters or figures depending upon how the line starts, to help ensure that the machines have time to get to the start of a new line. Again, make your tapes as short as possible by taking out any unneeded characters:

extra letters, upshifts followed by downshifts, and things like extra spaces or downshifts at the end of a line. Above all, be sure to put your credit line at the end, with the hope that others will follow your lead and keep it there.

So if RTTY pictures are your interest, try your hand and make a few. Your RTTY friends will be pleased to receive them.

CHAPTER 12

Improving Reception



It is a big thrill to watch your printer pounding out perfect copy from a distant RTTY station—but it is far from thrilling when bursts of interference suddenly cause the copy to look like a cartoonist's idea of profanity! But don't give up. There are several things you can do to improve this situation.

Most RTTY receiving converters convert the incoming FSK to audio tones. These audio tones are then passed through a limiter and then to tuned filters with associated tone detectors. This method of FSK detection is fundamentally the same as FM and exhibits much the same characteristics as FM. Whenever the signal we are trying to copy is stronger than interfering signals, the "capture effect" occurs. The desired signal passes through the limiter and the interference is attenuated. So—to obtain a big reduction in the effects of interference on our RTTY copy, it is only necessary to reduce the interference amplitude to less than that of the signals we want.

There are three points in the average receiving system which can usually be improved significantly. These are:

(1) The IF amplifier
(2) The second detector
(3) The audio ahead of the RTTY converter

Your particular receiver may be ideal in one or more of these areas. You should be able to judge from the following discussion whether you will benefit from some modifications in your receiving setup.

IF Amplifier Considerations

As in any communications system, the bandwidth of the IF amplifier should be no greater than necessary to pass the spectrum of the signal you want to copy. Fig. 12-1 illustrates the spectrum of an FSK signal using 850 Hz shift. Notice that practically all of its energy is contained in a 1200 Hz bandwidth. So, we see that 1.2 Hz is the optimum IF bandwidth for amateur RTTY reception.

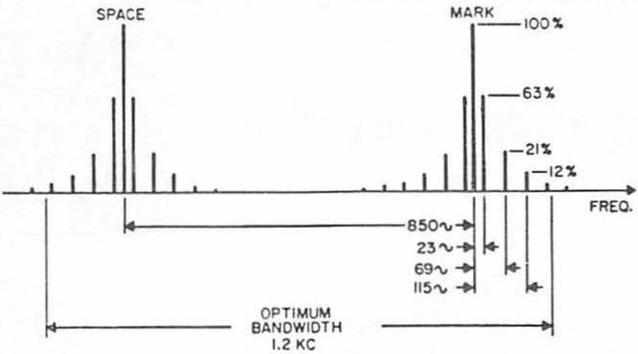


Fig. 12-1. Spectrum of 850-Hz shift FSK signal.

Many old receivers have IF bandwidths designed for AM signals 6-10 kHz wide. If you are using one of these for RTTY you should definitely take steps to improve the situation. To illustrate how effective using the correct IF bandwidth is, take a look at Fig. 12-2. Here a receiver designed for AM reception is shown having an IF bandwidth of about 6 kHz at the half power points (-3 db). A CW signal which is 6 db (one S-unit) stronger than our FSK signal comes on about a kilohertz away. Since it is passed on to the RTTY converter, it will capture the limiter and you start printing gibberish. Now, you

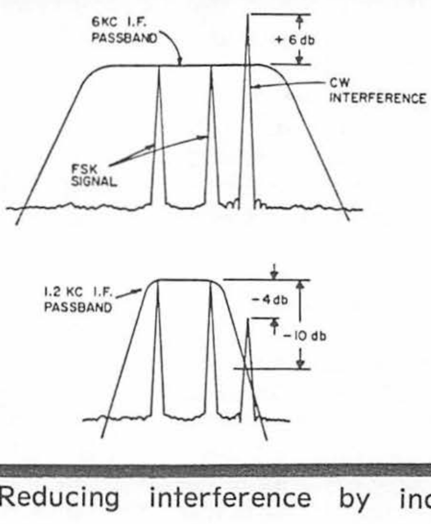


Fig. 12-2. Reducing interference by increasing IF selectivity.

can narrow the IF down ot 1.2 kHz as shown. The skirts of the filter attenuate the CW signal by 10 db; so it is now 4 db weaker at the converter than our FSK signal and our FSK signal has the upper hand. Perfect copy again!

Many modern receivers designed for SSB use have good mechanical or crystal lattice filters with nice, steep skirts and bandwidths of 2 kHz or slightly higher. These are excellent for RTTY, and optimum audio filtering (to be described later) is all that is required to put the finishing touches on the receiving system. If you have one of the older receivers which has an excessive IF bandwidth, there are several possible ways of improving the situation. Obviously, a mechanical or lattice filter of the desired bandwidth can be installed or added by means of an adapter.

If your receiver has a conventional crystal filter, some improvement can be obtained if care is taken in adjusting the filter. The selectivity setting and phasing control should be experimented with to find the best settings and these settings marked on the panel for easy resetting. You can easily get the filter adjusted too sharply which chops off part of the FSK spectrum. This will cause the mark and space tones to have unequal amplitudes.

The ubiquitous Q-fiver is also a good solution to the broad receiver problem. The old BC-453 has an ideal passband for RTTY when the IF coupling rods are pulled all the way out.

Second Detector

At first glance, it would seem that the second detector would not be a fruitful point for improvement. However, if your receiver has a conventional diode detector, a worthwhile gain in RTTY performance can be obtained by changing to a product detector. The reasons are much the same as for using a product detector for SSB. A diode detector produces beat-notes between all signals. Thus, if there are interfering signals coming in with the FSK, the diode detector will generate new frequencies from all of those signals beating together. The result is an increase in noise. This just gives the RTTY converter that much of a harder job. When a product detector is used, it is linear with respect to the signal input and only beats between incoming signals and the bfo appear at its output. So, we give our RTTY converter the cleanest possible signal. If your receiver doesn't have a product detector, you should certainly install one. It will help your SSB and CW reception also. Fig. 12-3 shows simple circuits which can be adapted to most receivers.

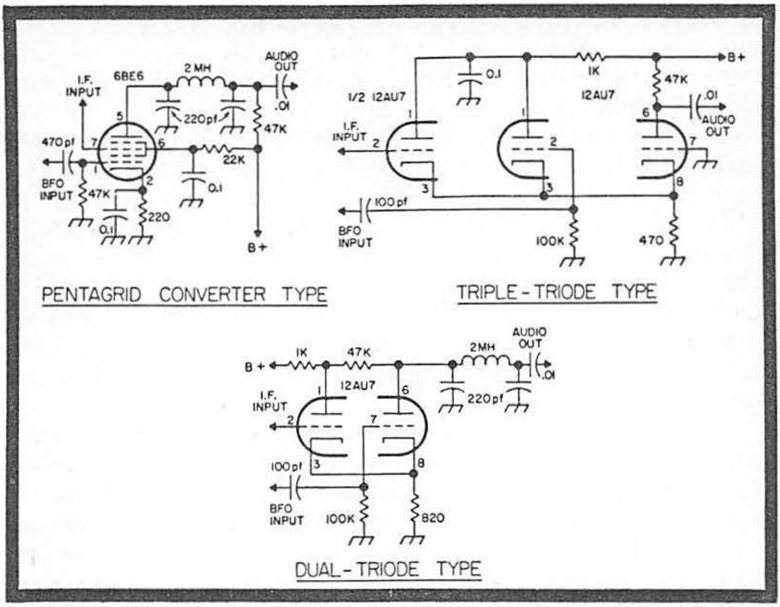


Fig. 12-3. Product detector circuits that can easily be adapted to most communications receivers.

Audio Filtering

One of the easiest and most effective ways of improving RTTY reception is by the construction of a double bandpass filter to be used ahead of the RTTY converter. If you include a switch to cut the filter in and out, you can demonstrate very dramatically how the filter can allow perfect copy even with heavy interference.

Looking back at Fig. 12-1 we can see that the FSK energy is concentrated about the mark and space frequencies. The filter to be described has a response curve as shown in Fig. 12-4. Note that the response is down over 20 db halfway between mark and space. Thus, we can have a CW signal at 2500 Hz nearly 100 times as strong as our RTTY signal and still get good copy. The half-power bandwidth of each bandpass section is about 250 Hz which allows some margin for misadjusted shifts and for a small amount of drift before retuning of the receiver is necessary.

The circuit of the complete filter amplifier is shown in Fig. 12-5. A cathode follower V1 is used to drive the two bandpass filters in parallel. The two series resistors are essential to provide the correct driving impedance since the filters are modified m-derived types requiring a match at input and output.

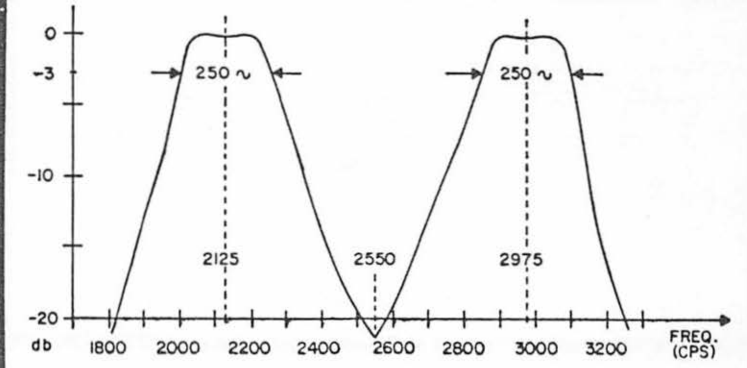


Fig. 12-4. Response of audio mark-space filter.

They are also terminated with the proper load impedance as shown. V2 is used as a simple combiner and to provide some gain to overcome the insertion loss of the filter sections. A switch is included to be able to bypass the filter when narrow shift is being copied. The input gain control can be permanently set to give unity gain through the filter-amplifier, if extra gain is not needed.

Details on building the filters are given in Figs. 12-6 and 12-7. The inductors are 88 mh surplus telephone loading coils which are widely available. The capacitors used should be paper, mylar, or mica. The ceramic types are often voltage-

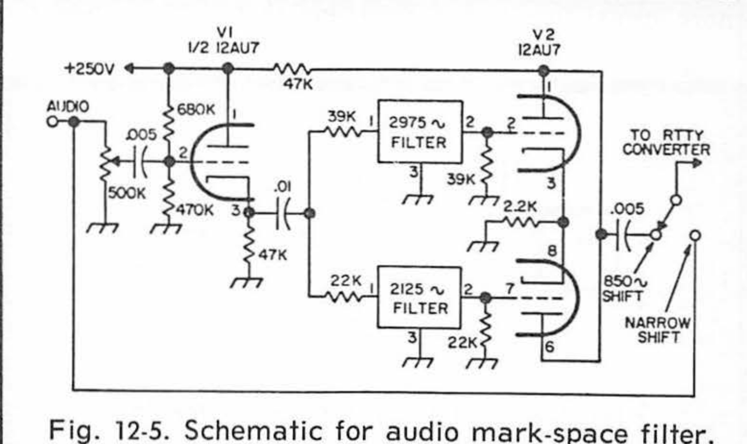
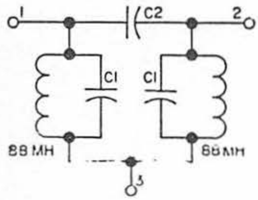


Fig. 12-5. Schematic for audio mark-space filter.



CENTER FREQ.	C_1	C_2	TUNE EACH L-C CIRCUIT TO:	LOAD "R"
2975~	.03*	.0022	3075~	39K
2125~	.06*	.005	2225~	22K

Fig. 12-6. Circuit and values for sharp filters.

sensitive so are not recommended. Notice in Fig. 12-6 that each toroid must be peaked at a frequency slightly higher than the filter center frequency. Fig. 12-7 shows a test set-up for tuning. A calibrated audio oscillator and a VTVM or scope is needed. If you don't have an audio oscillator, an LM or BC-221 frequency meter can be used. Set the meter up on 2000 kHz with the crystal calibrator on. You can then obtain accurate audio tones from the headphone jack by detuning the meter dial. For example, if you want 2125 Hz, just look in the calibration book for 2002.125 Hz and reset the main dial to that reading. The beat note with the crystal will then be 2125 Hz.

With the circuit as shown in Fig. 12-7, select a capacitor for C1, and tune the audio oscillator until you get a peak on the meter or scope. If the audio frequency is not the value desired, you can adjust the circuit by either changing the capacity or removing turns from the toroid. If you wish to remove turns, then use a capacitor that gives a resonant

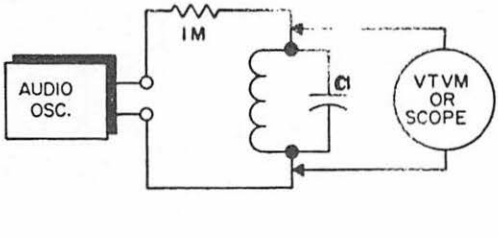


Fig. 12-7. Test setup for tuning toroid.

frequency slightly lower than shown in Fig. 12-6. Then unwind turns until you hit the specified frequency. If you don't want to fool with unwinding the toroid, then select a capacitor which gives a slightly higher reading. Then shunt it with small capacitors until the right frequency is obtained. Of course, if you are lucky, you may find one capacitor which hits the right frequency. Incidentally, small ceramics in the .001- to .003-mfd range are all right for trimming since small variations in these would have negligible effect.

When all four toroids are properly tuned the filter can be assembled. Small pieces of perforated board with flea clips are very handy for mounting the components. When installing the filters in the amplifier be sure to provide isolation between input and output to keep down leakage around the filters.

IMPROVING WEAK SIGNAL RECEPTION

The circuit in Fig. 12-8 is somewhat conventional, but with balanced detector, trigger and keyer tubes. A filter circuit consisting of an .02 mfd condenser, two 100K resistors and the 220K resistor form a long time constant circuit in the detector output which completely wipes out noise spikes that could trigger the keyer stage when signals are weak. Observations on the scope indicate all traces of noise spikes are eliminated by this circuit. It can be shown that removal of these condensers during noisy reception will cause very bad garbling of otherwise perfect copy. Note coupling resistors R1 and R2 on the grid of the driver tube. Both resistors are identical, their value being determined by the voltage developed at the plates of the 6AL5 detector. These are part of a voltage divider to drop the DC voltage on the grids of the 12AU7 to -8 volts as measured on a VTVM, when the respective half of the 6AL5 is conducting. Transformers T1 and T2 are approximately 1:1. If substitutions are made, it will only be necessary to adjust the coupling resistors R1 and R2 so that -8 volts appears on the 12AU7 grids when the limiter is saturated and with the proper tone at the converter input. The K1 balance control in the cathode of the 12AX7 amplifier is adjusted to equalize the voltages on the 12AU7 grids when the respective mark or space frequencies are selected. Minus 8 volts on the 12AU7 grid will cause an increase in the plate voltage to at least 60 volts, which will fire the neon properly.

The 6SN7 keyer tube could operate balanced with one winding of a type 255A polar relay in each cathode, but a better way is to replace one winding with a 150-ohm resistor and provide bias current to the disconnected winding. In this case,

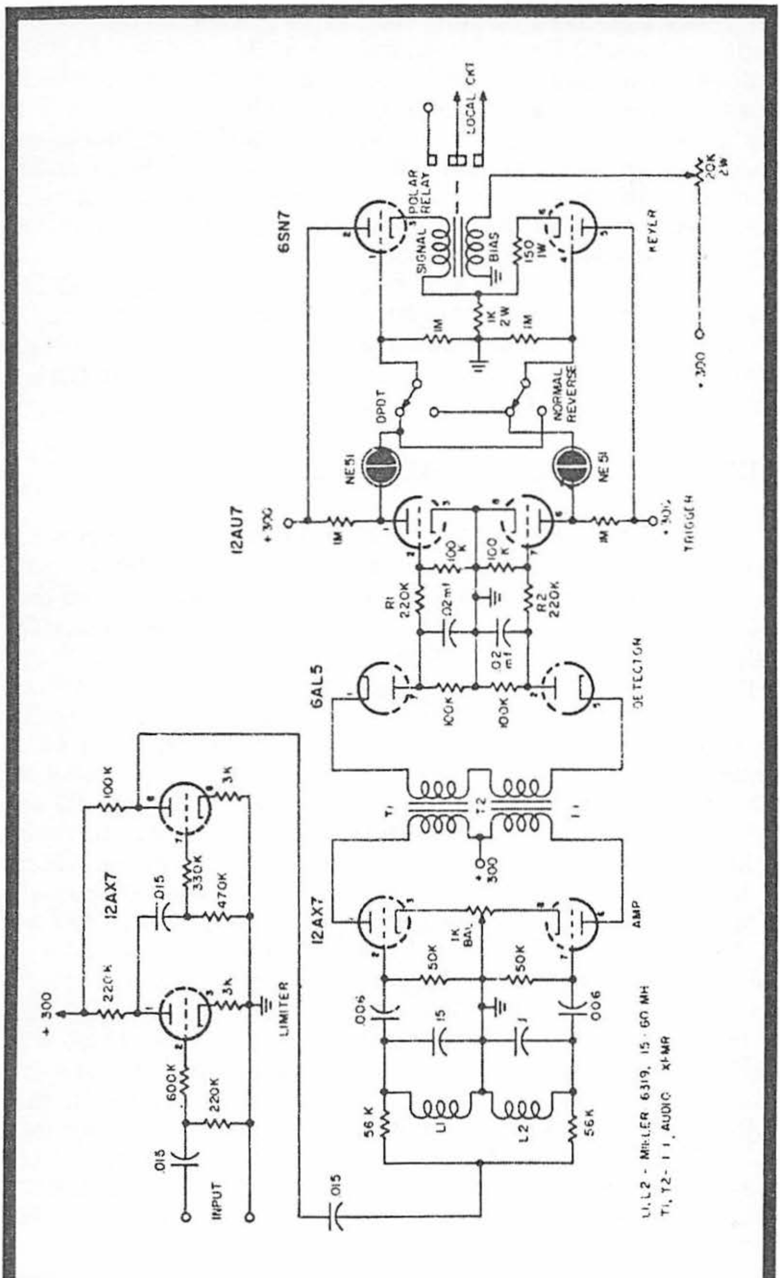


Fig. 12-8. Schematic of circuit for improving weak signal reception.

the actual keying is done on space signals, but you have a built-in mark. The polar relay bias current should be adjusted to exactly ½ the current value of the signal winding when that portion of the 6SN7 is conducting. In this case 30 ma in the signal winding necessitated a bias current of 15 ma.

Test Results

The DPDT switch on the NE-51's allow copy of normal or reversed signals. Prior to the development of this circuit I had tried several discriminator type detectors followed by various trigger and keyer circuits; but with signals down in the noise, the above converter would outcopy the discriminator type of converter. Comparison was direct, with the two converters side by side.

The TV width coils can be replaced with 88 mh toroids by tuning them to the proper frequencies. The toroids give somewhat better selectivity, and due to the higher Q there is an increase in the detector output voltage of approximately 8 or 9 times. With toroids it will be necessary to increase the value of R1 and R2 to 1.5 megohms.

As a matter of interest, selectivity curves were run on both the toroids and the TV width coils. The width of the selectivity curve at the ½ voltage point (6 db down), is as follows:

	2125 center freq.	2975 center freq.
TV width coil	570 Hz	890 Hz
88 mh toroid	180 Hz	330 Hz

It can easily be seen that toroids would be a definite improvement when the QRM is bad.

Another real problem in radio-teletype reception is selective fading. One does not notice this unless the audio is either metered or observed on a scope. This can be quite severe at times. I have observed as much as 20-db difference in the two tones, first one and then the other being the stronger. I might suggest the use of separate audio limiters in each channel to help overcome this. Tuned circuits should be used ahead of the limiters to separate the two tones, prior to limiting.

EQUALIZING AFSK TONES

Common methods of keying audio oscillators for AFSK lead to unequal mark and space amplitudes. The common method of obtaining an audio frequency shift signal is to

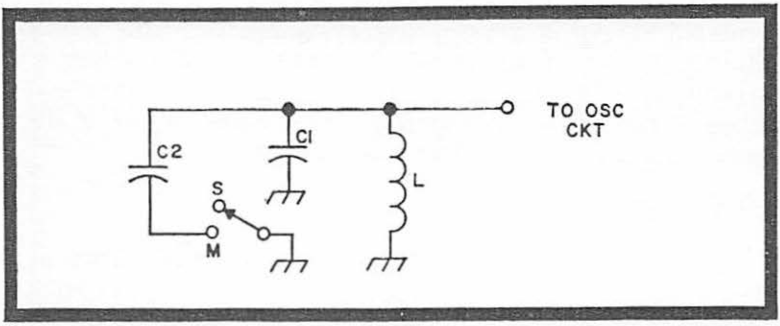


Fig. 12-9. Common method of obtaining an audio frequency shift for RTTY.

switch the tuning capacitance of an audio LC oscillator, as shown in Fig. 12-9. The space frequency (2975 Hz) is formed by LC, while the lower mark frequency (2125 Hz) is formed by L-(C1 + C2). While adding C2 lowers the frequency, it unfortunately lowers the circuit impedance which usually lowers the Mark output level, depending on the oscillator circuit and the circuit Q. This reduced mark output can cause distortion in the receiver TU, and lowers the mark signal-to-noise ratio.

This mark-space amplitude difference is either neglected or equalized by a C or LC network in the oscillator output. However, there is a much simpler method of tone equalization.

The Twin City AFSK circuit was used as a typical oscillator for tests, with the keying circuit temporarily omitted (Fig. 12-10). The mark amplitude was measured at 3 db below the space amplitude. Let us see what can be done about this.

Instead of thinking of the mark amplitude as being too low, let us consider the space amplitude to be too high, and look for an easy way to lower it. Adding resistor R1 (Fig. 12-10) across the switch does the trick. The mark circuit (LC2 C2) is not affected, but the space circuit now has its Q lowered by R1 in series with C2. As the value of R1 is decreased from a very high value the space amplitude drops quickly, but the space frequency is almost unaffected, as shown in Fig. 12-11. For this particular circuit the tone amplitudes will be equal if R1 is 125K. The space frequency shift due to R1 C2 is only 3½ Hz.

Now let us continue with the diode keying circuit of the Twin City circuit, Fig. 12-12. R1 loads the space circuit through the keying diodes, D1 and D2. The circuit characteristics were

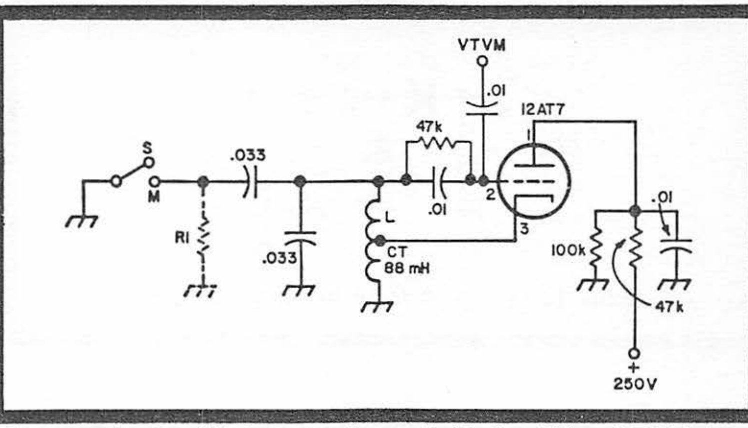


Fig. 12-10. Oscillator section of Twin City AFSK less keying diodes.

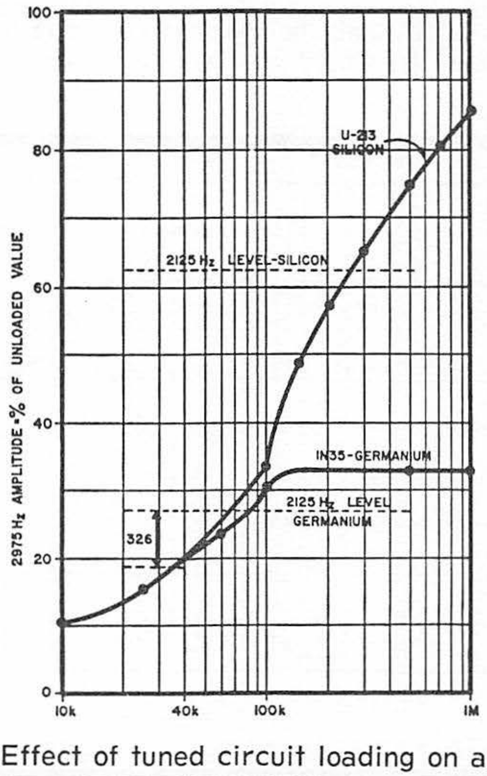


Fig. 12-11. Effect of tuned circuit loading on amplitude.

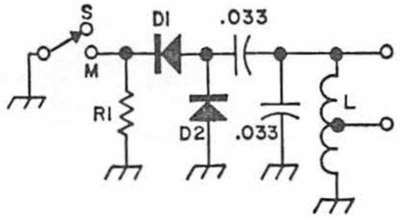


Fig. 12-12. Twin City keying circuit.

first measured using silicon TV diodes (Texas Instruments U-213) for D1 and D2, with the results shown in Fig. 12-13. Note that the mark and space tones will be equal in amplitude if R1 is 250K.

Next, germanium 1N35 diodes were tried. These are similar to the Twin City 1N54's. Fig. 12-13 shows that the germanium diodes load the space circuit, due to their low back-resistance. R1 should be about 85K for equalization, but

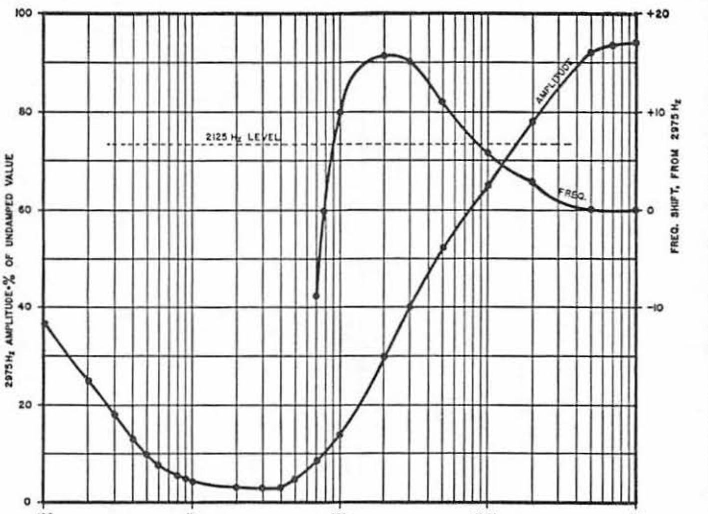


Fig. 12-13. Effects of diode loading on space amplitude.

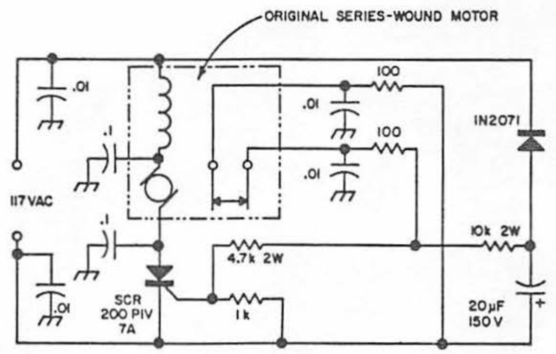


Fig.12-14. Using a SCR in a teletype series-wound motor.

this will vary with the particular diodes used. (The voltage across the oscillator circuit is a bit high for 1N54's and 1N35's.) If R1 is 39K, as shown in the Twin City circuit, the space amplitude will be about 3 db below the mark, just the opposite of the usual situation.

While the proper value of R1 will vary with the circuit used, an equalizing value can be found for almost any oscillator and keying circuit. Remember that equal mark-space percentages of modulation are the goal. If your speech amplifier is not flat, this system will permit the proper amplitude adjustments.

MOTOR NOISE REDUCTION

This circuit (Fig. 12-14) may be used for greatly reducing the spark and resulting interference from motors that are commonly found in a Teletype machine.

The resulting current across the governor contacts is reduced from about 1 ampere down to about 10 ma. The noise generated becomes completely inaudible.

When there is no voltage coming into the gate of the SCR, the motor speeds up. When there is a slight negative voltage coming into the gate, the motor slows down. Using this theory, you can make up a little negative supply: the 1N2071 and the 20 mfd, 150-volt condenser. The output of this is then applied to a voltage divider consisting of 14.7K (10K and 4.7K), and the 1K resistor between the gate and the cathode of the SCR.

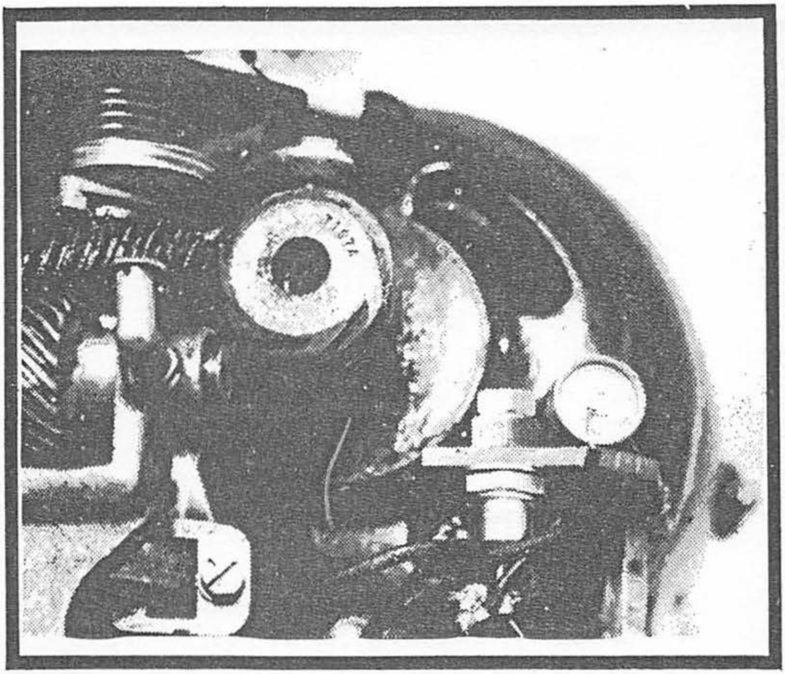


Fig. 12-15. Rear view of a Model 14 Teletype with the SCR mounted at the lower right. It is mounted on the small bakelite board with the condenser on the top.

When the motor is either starting up or going too slowly, the governor contacts are closed, which makes the SCR receive no negative voltage, thus giving the motor more voltage. When the motor has attained its proper speed, the contacts open, and the SCR's gate receives a slight negative voltage, thereby decreasing the speed of the motor.

You may find that you can further reduce any interference which may show up by substituting small 2.5 mh chokes in place of the 100-ohm resistors. For the perfectionists, put an Ohmite Z-7 choke, bypassed to ground, in each lead coming out of the motor.

RANGE ADJUSTMENT

Most printers have a range adjustment, calibrated from zero to one hundred. If you adjust the range while copying an incoming signal you can see how far you can go in either direction before you start errors in your copy. Optimum is perfect copy from 10 to 90, as 80 point range. Errors will be at a

minimum under these conditions if you set the range at 50. If the transmitted signal favors either the mark or space frequencies you will find the range swinging to one side or the other. A range of 30 to 100 would indicate a 20 percent spacing bias, which would probably be due to an incorrectly adjusted polar relay somewhere along the line. If printer speed is slower than normal you will notice that both margins are raised, the lower margin going up more than the upper. Vice-versa for higher printer speeds.

CHAPTER 13

Filters

Until very recently filter design was essentially a cut-and-try proposition. Classical theory upon which the design was based called for physically impossible components in the filter, and substitution of realizable items led to inaccuracies in the design. This situation, fortunately, has now been cured.

CLASSES OF FILTERS

The purpose of a filter is to separate AC signals. In general, a filter must fall into one of four categories: highpass, lowpass, bandpass, or bandstop. A highpass filter passes all signals higher in frequency than its cutoff frequency and stops all lower-frequency signals. A lowpass filter does the reverse. A bandpass filter passes all signals between its lower and upper cutoff frequencies and blocks signals either higher or lower in frequency than its passband limits, and a bandstop filter blocks passage of frequencies within its band while permitting all others to pass.

These four categories of filters are based upon the action performed by the filter. Any filter, though, must operate at some specific frequency or frequencies. The frequencies involved may be subsonic, audio, intermediate, or radio frequencies. The design of any specific filter depends upon both the action to be performed, and the operating frequencies involved. Components employed in the filter, and the apparent principle of operation, may vary widely with the action and the frequency.

FILTERING ACTION

However, when we get right down to the basics we find that all filters operate on the same basic principles despite apparent differences. These apparent differences are best illustrated by some examples of various filters: The filter portion of a power supply is a lowpass filter operating in the

subsonic frequency range, while the TVI filter on a transmitter output is usually a lowpass filter operating in the RF region. A TVI filter of the sort connected to affected TV receivers is a highpass filter operating in the RF range. The mechanical filter used in SSB work is a bandpass filter operating at the intermediate frequency, and so are the crystal lattice filters also used in SSB. For that matter, an IF transformer is a bandpass filter, as is any resonant circuit. RTTY filters are also bandpass designs, but operate in the AF region.

All filters are based on the frequency-dependent properties of reactances (even the mechanical filter makes use of the mechanical equivalent of electrical reactance). Capacitive reactance, Xc decreases as signal frequency f rises, while inductive reactance XL goes up with increasing frequency. By proper choice of the type and amount of reactance any desired filter action can be obtained (within reasonable limits, as we shall see a little later).

A Simple Filter

To illustrate the principle, let us examine a single stage, subsonic, lowpass filter typical of those used in many power-supply designs. The schematic appears in Fig. 13-1.

At frequencies far below the cutoff frequency of this stage, the inductor presents a very low value of XL and the capacitor has a very high value of XC. Both reactances are so extreme that they may be ignored, and the filter becomes in effect a straight piece of wire with no effect on the signal.

At frequencies far above cutoff, the situation reverses. The inductor's reactance is very great and the capacitor's

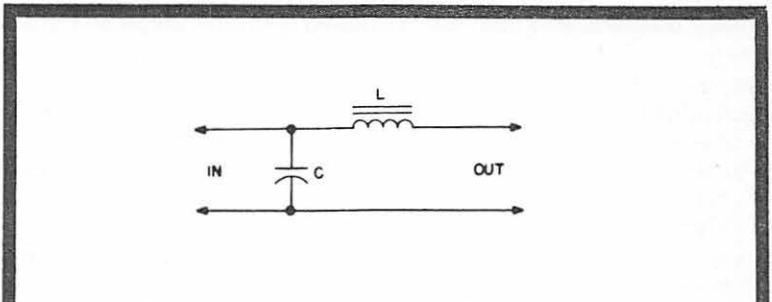


Fig. 13-1. Single section low-pass filter. Values of L and C determine cutoff frequency of filter section.

reactance is very low. The filter becomes in effect a dead short, with no path from input to output, and the signal is blocked.

The "cutoff" frequency of this type of filter is defined as being that frequency at which XL equals XC. At this frequency, since reactance is equal in both directions, half the signal current flows to ground and the other half flows through the filter. The signal is neither passed without loss nor blocked completely.

At frequencies near cutoff, the situation is similar to that at the cutoff frequency. Signals slightly below cutoff frequency will be slightly attenuated, but the lower the frequency becomes the more of the signal goes through and the less of the signal is bypassed to ground. Signals slightly above cutoff are still partially passed, but, as the frequency rises, less and less of the signal makes it through the filter. There is no such thing as a perfect filter which passes every thing below cutoff and blocks all above.

Cascading

Filter performance can be improved, though, by adding more stages. If two of these stages are cascaded one after the other, then at cutoff frequency the first stage will block half the signal and permit half to pass. The second stage will block half of the half that passes through the first, and permit only ¼ of the original signal to get through the composite filter.

Adding a third such stage will reduce the output signal level to ⅛ that at the input. A fourth stage will have this, to one sixteenth the original level. Five stages will cut the output to one thirty-second, and so forth.

When extra stages are added, though, the definition of cutoff frequency must be changed. Each stage's cutoff frequency is defined exactly as before, but the cutoff frequency of the filter as a composite unit becomes lower with each added stage, because for any filter the definition of cutoff frequency most generally used is: "that frequency at which the filter reduces output signal level to half the level present at the input."

Our two-stage filter in the example would have a composite cutoff frequency only 0.7071 times as great as the single-stage filter; the three-stage filters cutoff would be at a frequency half that of the single stage filter. These ratios apply only to the simple design shown in Fig. 13-2; more complex filter designs have different rates of cutoff-frequency change, and in general a filter is designed for specified per-

formance rather than being built up of some arbitrary number of identical stages.

Highpass Filter

We have just examined the basic principles as applied to a simple lowpass filter. A highpass filter works the same, except that the inductor and capacitor are interchanged so that signals above cutoff are passed and those below cutoff are blocked rather than vice versa as in the lowpass circuit.

Bandpass Filters

To get a bandpass filter, we could simply build a highpass filter with cutoff set at the lower band limit, and follow it by a lowpass filter with cutoff set at the upper band limit. For extremely wide passbands this is sometimes done. One example is the 300-3000 Hz band frequently used for voice communication; this is too wide a bandpass for simple bandpass filters to handle, and the economical way out of the problem is to first trim off one limit and then trim the other.

For narrow passbands, though, something more like a tuned circuit is generally used.

Not all filter stages are as simple as that shown in Fig. 13-1. Tuned circuits may be included instead of simple reac-

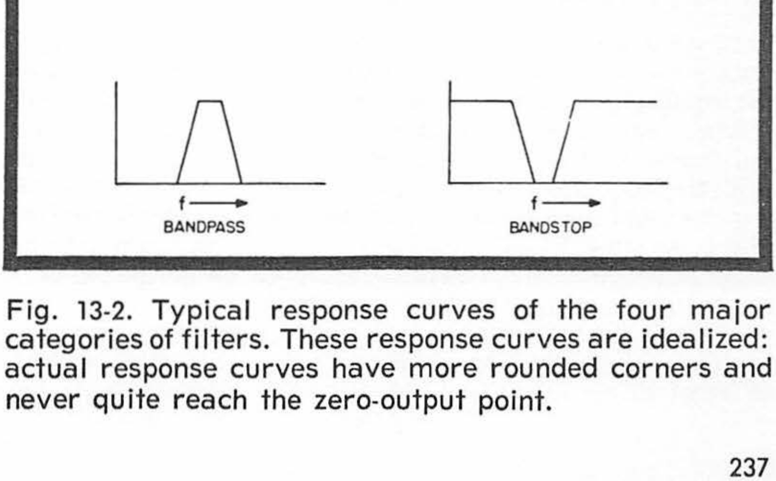


Fig. 13-2. Typical response curves of the four major categories of filters. These response curves are idealized: actual response curves have more rounded corners and never quite reach the zero-output point.

tances, and the resonance frequencies may be far different from the intended cutoff frequency in order to modify filter performance. The variations possible in filter design on this basis gave rise to the image-parameter theory of filter design, which ruled the design of most filters for more than 30 years and even today is still widely used.

However, reactances which satisfy the demands of image-parameter theory are physically impossible, because the theory demands that the resistive portion of the circuit be zero at cutoff frequency, changing in the same manner as a capacitive reactance below cutoff, and changing in the same manner as an inductive reactance above cutoff, while actual resistances are independent of frequency.

Because many filter requirements are only approximate, this design method has been able to give circuits acceptable for most purposes. When more accurate designs were required, cut-and-try fitting of values was the normal course.

Kirchoff's Laws

Any filter, though, is a network of components connected together, and any such electrical network must obey certain basic rules known to engineers as Kirchoff's Laws. These state, in essence, that as much current must flow into any point as flows out and vice versa. We might say simply that the current must go somewhere it cannot just disappear!

These rules can be written as a series of algebraic equations, and to apply them to the analysis of any arbitrary network we must do just that. The laws state that all the current must be accounted for, and the equations do the accounting.

MODERN NETWORK THEORY

Since a filter is a network, the filter can be described by a suitable series of equations. The filter designer can then solve the equations to find the network parameters which will produce optimum performance in some desired respect. Performance will be poorer than optimum in some other respects when this is done, since all design represents compromises, but the modern network theory design technique for filters permits the designer to choose the performance he wants where the image parameter approach permitted only one or two parameters to be designed for.

The modern-network-theory design approach permits a number of different filter response curves, and the resulting different filters are usually identified by names associated

with the shapes of these curves. The two most widely known are the Chebishev (also spelled Tshebysheff, Chevishef, and Tshebishev) and the Butterworth filters. The Chebishev design provides the sharpest possible rate of cutoff at the cost of permitting ripple in the passband, while the Butterworth design maintains passband response flat at the cost of moderate phase shift and less steep cutoff rates.

It may seem a bit odd for us to declare that all filters are based upon reactance when crystal lattice filters composed of several quartz crystals are so widely used, but electrically the crystal is just a very high-Q tuned circuit and contains both XL and XC. When used in a filter, the XL and XC provide the filtering action.

Filter Performance Rating

Filter performance is rated according to several factors, and the factors involved vary to some degree with the type of filter in question. One factor present in the rating of every filter is its attenuation; this is a measure of the signal level at the output side of the filter compared to the level at the input side.

Attenuation Versus Frequency

Every filter possesses some definite value of attenuation at every frequency. A low-pass filter has attenuation both below and above its cutoff frequency, as do highpass filters, bandpass, and bandstop varieties. Attenuation is usually expressed in db.

If the output level at some specified frequency is the same as the input level, the attenuation at that frequency is O-db. If the output level is zero for any values of input level (a condition theoretically possible but not attainable in practice because of stray coupling around the filter), the attenuation at that frequency is infinite.

A plot of attenuation versus frequency gives the response curve of the filter. Fig. 13-2 shows typical response curves for the tour categories of filters.

Cutoff Frequency

Another factor in common for all filters is cutoff frequency, but now that we have established the meaning of attenuation we are in position to define cutoff frequency more accurately than we did earlier. The cutoff frequency for a given attenuation may be specified as the frequency at which the filter has that value of attenuation. Thus, the 3 db cutoff

frequency for a low-pass filter will be lower than the 6 db cutoff frequency. The definition we developed earlier: "that frequency at which output level is half of input level," actually defines the 6 db cutoff frequency of any filter.

A glance at Fig. 13-2 will show that both bandpass and bandstop filters have not one but two cutoff frequencies, since for any specified value of attenuation there is a frequency below the filter's center frequency which has that attenuation, and another frequency above center frequency. The two cutoff frequencies of band filters are designated as upper and lower cutoff frequencies.

DB Per Octave

Sometimes filters are compared according to their slope or cutoff rate. This is a relative term referring to the rate at which attenuation changes with frequency. Engineers speak of slope or cutoff rate in units of db per octave; this means the number of db change in attenuation when the frequency is doubled. A single reactance has a slope of 6 db per octave; that is, if frequency is doubled the reactance is either doubled or halved, and the resulting ratio is 6 db (2:1 voltage or current, and 4:1 power ratio). Filters with two reactances usually have slopes of 12 db per octave, with the slope increasing 6 db per octave for each added reactance in the filter. This is not an airtight law, however; many filters have slopes which do not follow this rule, and almost no filter follows this rule at frequencies near the cutoff point or within the passband.

Bandwidth

All filters have a bandwidth, but the term has different meanings depending on whether the filter is (1) low-pass or high-pass, or (2) bandpass or bandstop. For low-pass or high-pass filters, the bandwidth for a specified attenuation is the actual frequency at which the filter has that attenuation. The width means the number of cycles from zero frequency to the frequency in question. Thus the bandwidth for 3 db attenuation of a low-pass filter with a 3 db cutoff frequency of 100 Hz is 100 Hz.

For symmetrical bandpass or bandstop filters, the bandwidth is the difference (in hertz) between the two cutoff points for the specified attenuation. A bandpass filter with an upper 3 db cutoff frequency of 10 kHz and a lower 3 db cutoff frequency of 8 kHz would have a 3 db bandwidth of 10 minus 8 or 2 kHz.

Bandwidth is always specified for a definite attenuation level, because it is defined in terms of cutoff frequencies, which are themselves defined only by attenuation level. Unless the attenuation level is specified, any use of bandwidth as a rating factor is meaningless.

For bandpass filters in particular, the bandwidth is one of the primary measurements, but it is by no means the only one. Consider two different bandpass filters, both having 6 db bandwidths of 3000 Hz. The first, however, is a very simple filter with only a few reactances and consequently has a very slow rate of cutoff, while the other is a complex Chebishev design with steepest possible cutoff rate. The first filter may have a 60 db bandwidth of 30 kHz: the 60 db bandwidth of the second might be as narrow as 6 kHz (Fig. 13-3). Obviously, the second of these filters will do a far better job of trimming applied signals down to its passband.

Shape Factor

The term shape factor is used to describe the property we have just compared. The shape referred to is that of the filter's response curve. A perfect bandpass filter would have absolutely vertical sides to its response curve, with zero attenuation between the two cutoff frequencies and infinite attenuation outside that range. Actual bandpass filters have

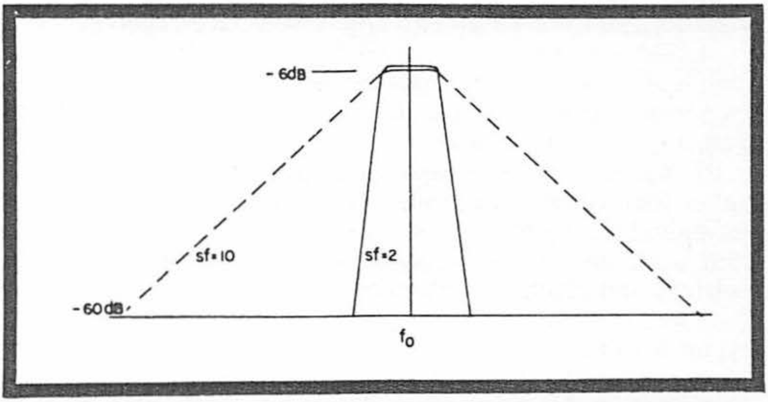


Fig. 13-3. Significance of shape factor is shown here. Both filters whose response curves are illustrated have 6 db bandwidths of 3 kHz. That in solid lines has a 60 db bandwidth of only 6 kHz, while that in dotted lines has 60 db bandwidth of 30 kHz. Difference in effective passband is clearly evident.

response curve which vary from a barely perceptible hump up to a steep-sided curve very much like that of the perfect filter.

The shape factor of a bandpass filter is the ratio between its bandwidths at two different attenuation levels. In ham practice, the 6 db and 60 db bandwidths are the ones most often used. The shape factor ratio is always obtained by dividing the larger bandwidth by the smaller, so that it is always greater than 1. A perfect filter with vertical sides to the response curve would have the same bandwidth at all attenuation levels, and so would have a shape factor of 1. Practical filters always have greater bandwidth at high attenuation levels than they do at the lower-attenuation points, and so their shape factors are always greater than 1. If the 6 db bandwidth of a filter is 1.5 kHz and the 60 db bandwidth is 3.0 kHz, the 6-60 db shape factor of that filter is 3.0 divided by 1.5 or 2; this represents excellent performance. Many bandpass filters in use today have shape factors as great as 10. The shape factor of a single tuned circuit when used as a bandpass filter may be as great as 100.

HOW ARE RF BANDPASS FILTERS CONSTRUCTED?

One of the requirements for any receiver intended to operate in today's crowded RF spectrum is that it have extreme selectivity. Ideally, it should be able to tune in either sideband of an AM signal while rejecting both the carrier and the other sideband. This is known as selectable sideband capability, and marks an extremely selective receiver.

Almost universally, such selectivity is achieved by use of bandpass filters operating in the RF or IF range. Since normal IF's are actually radio frequencies, we'll lump these filters together for our discussion.

RF bandpass filters capable of providing shape factors smaller than about 10 are most often one of two major types: mechanical filters and crystal filters. We won't go into details of the mechanical filters here, since they have little to do with electrical and electronic phenomena.

Crystal Filters

Crystal filters, themselves, divide into two major groups. One of them uses only a single crystal, to provide a very narrow response curve with relatively poor shape factor. This type, which was standard equipment on older communications receivers, provides excellent results on CW but is not so good for reception of voice signals. The second group, which used two or more crystals, provides greater bandwidth than the

first, with smaller shape factor. It is widely used in SSB operation, since the shape of its response curve matches the requirements for voice communication.

The single-crystal type of filter is normally known simply as a crystal filter, while those using two or more crystals are generally called crystal lattice filters. Technically, a lattice filter must use four crystals, or multiples of four, but an electrically equivalent circuit called a half-lattice requires only two crystals, and is possibly the most popular type of lattice filter in use now.

Let us examine both groups of filters separately, taking the older single-crystal filter first. The schematic of a typical single-crystal filter circuit appears in Fig. 13-4. The driving transformer splits the signal into a pair of signals, equal in strength but opposite in phase. One of these two signals is applied to the quartz crystal while the other is applied to a phasing condenser which is simply a variable capacitor. The two signals are then put back together at the input to an amplifier stage.

The quartz crystal is equivalent to a very high-Q resonant circuit; it has both series and parallel resonances. At the frequency of series resonance, it is an extremely low resistance, while at the frequency of parallel resonance it is an extremely high resistance. In the absence of the phasing condenser, this circuit would permit signals at series resonance to pass through almost unchanged, while attenuating signals at other frequencies greatly and providing almost infinite attenuation at parallel resonance.

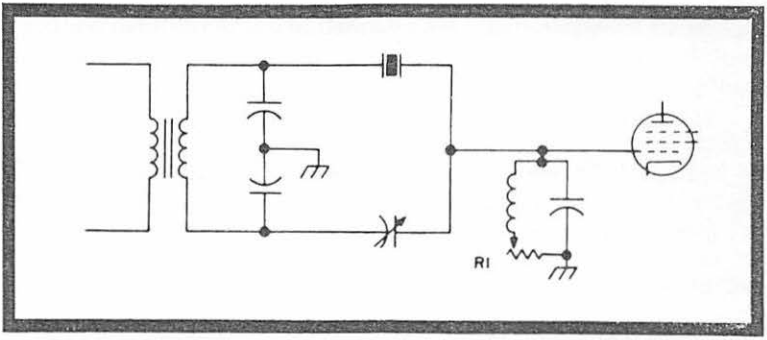


Fig. 13-4. Typical single-crystal IF filter of type used for CW reception. Adjustment of capacitor tunes out parallel resonance of crystal to give symmetrical passband, or can move parallel resonance to either side of series resonance to insert a "rejection notch" anywhere within passband except at series resonant frequency itself.

Phasing Condenser

The phasing condenser, however, provides an alternate path for the signal around the crystal. When the phasing condenser is adjusted so that its capacitor exactly balances the parallel capacitance of the crystal and its holder, the parallel resonance of the crystal is eliminated and the filter's response curve becomes a single sharp peak. The rejection notch at parallel resonance is eliminated.

When the phasing condenser is adjusted for either more or less capacitance than that required to balance the crystal, the effect is to tune the parallel resonance of the crystal to some definite frequency. This introduces the rejection notch, and makes it possible to move the rejection notch relative to the passband peak, permitting a single interfering frequency to be notched out of the signal.

Adjustment of resistance R1 modifies the loading imposed on the filter, and varies the effective bandwidth. The 6 db bandwidth of such a filter circuit can be varied from about 50 Hz when R1 is extremely large, out to greater than 6 kHz when R1 is very small.

Multiple-Crystal Filters

The multiple-crystal filter designs are based on the lattice structure shown in Fig. 13-5. Each section of this lattice structure consists of one or more crystals, and provides the equivalent of the circuit shown in Fig. 13-6. Much of this circuitry can be eliminated by replacing the lattice with its equivalent shown in Fig. 13-7, the lattice arms themselves now require only single quartz crystals using their series resonances.

Characteristics of such a filter are determined primarily by the number of crystals used, and the degree to which they are matched. Those crystals marked A and those marked B must not tune to the same frequency. In fact, the spacing between the series resonant frequency of crystals A and that of crystals B is the major factor determining the filter's bandwidth.

Chebishev Techniques

Design of such a filter is done by modern network theory, usually using the Chebishev techniques, and is too complex for us to explore here. In general, the bandwidth determines the spacing between resonant frequencies of the two groups of crystals. Within each group, characteristics of each crystal

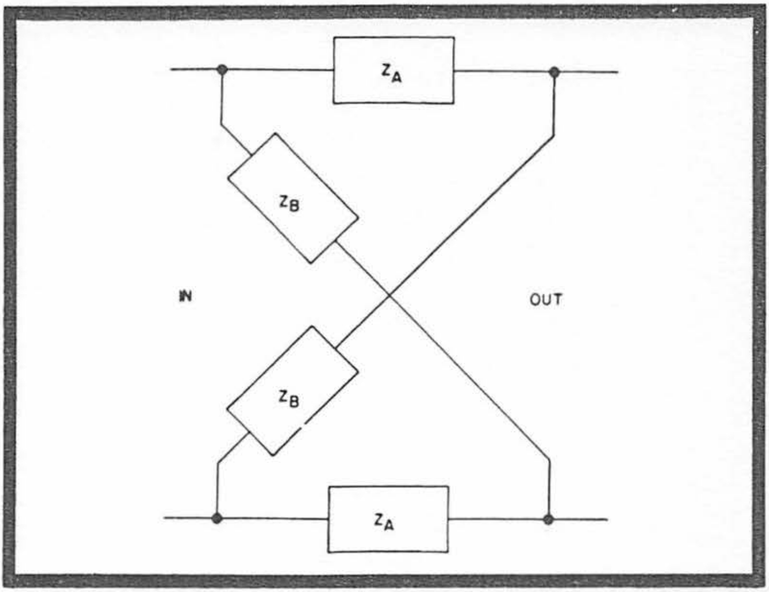


Fig. 13-5. Full-lattice filter arrangement in block diagram form. Each block represents an impedance having characteristics of the circuit shown in Fig. 13-6. When interconnected as shown here, passbands of A and B sections interact to provide a flat-topped passband with excellent shape factor.

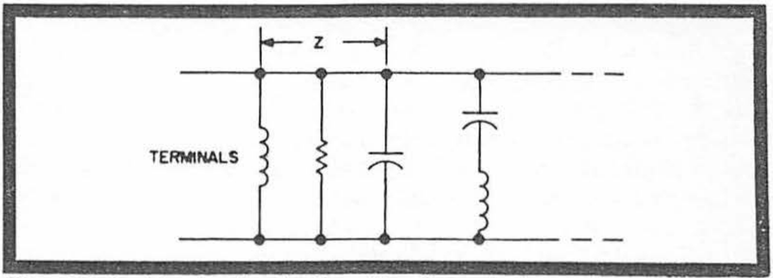


Fig. 13-6. Equivalent circuit of each impedance in full lattice filter. Parallel resonant circuit at left represents parallel resonance composed of crystal and stray circuit capacitance, together with loading resistances. Series-resonant circuit at right (which may be extended indefinitely as indicated by dottled lines) represents series resonances of each crystal in each leg. Impedances A and B are parallel resonant at lower cutoff frequency. Series resonances introduce infinite rejection notches just outside passband to sharpen skirts of filter.

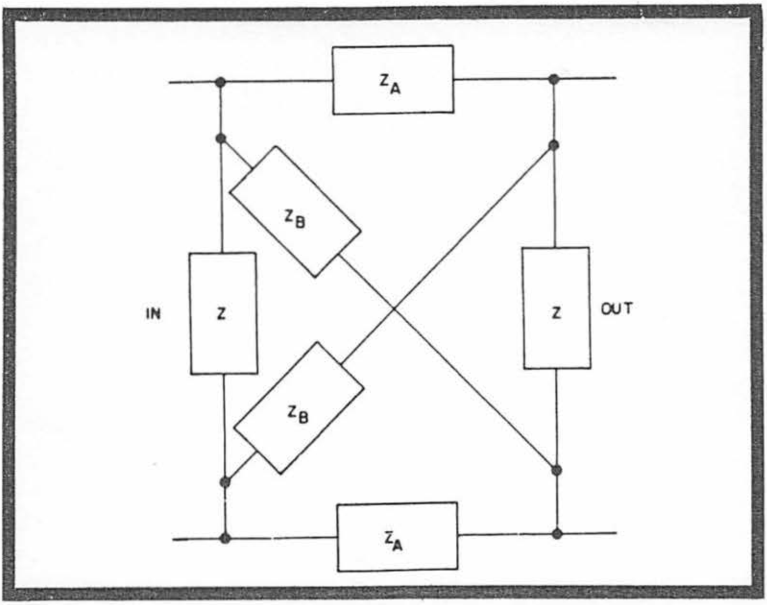


Fig. 13-7. Load resistances and stray circuit capacitance of full lattice filter can be moved to input and output terminals as indicated here. This circuit is electrically the same as that of Fig. 13-5. Each A and B block is still the circuit of Fig. 13-6, which may in practice consist of a single crystal for each block.

must match as closely as possible. Many commercial crystal filters use a single crystal with multiple electrodes on it. Each pair of such electrodes serving as a different crystal in the lattice, to assure identical characteristics. So long as the filter's center frequency is not greater than about 250 times its desired bandwidth, good shape factor can be attained easily. That is, for a phone bandwidth of 3 kHz, any center frequency up to 750 kHz could be employed. In fact, successful lattice filters with center frequencies as high as 10 MHz and bandwidths as small as 2 kHz have been built, but design is more critical.

DISCRIMINATORS

Teleprinter operation by amateurs usually consists of the transmission of two frequencies, one separated from the other by 850 cycles. One frequency represents the mark signal and the other the space signal. With frequency shift keying (FSK)

normal practice is to have the lower of the two frequencies as the space signal and the higher frequency the mark signal. The problem in reception is, briefly stated, to use specialized receiving equipment which, when the frequency representing the mark signal is received, provides current to the selector magnet of the machine and when the space signal is received cuts off the flow of current to this magnet. A filter (or a discriminator, which is basically a specialized form of a filter) must be used for separating this information. Our objective here will be to give some indication of the considerations that go into the choice of filter, and some help in getting a reasonably adequate filter operating.

MAB Keying

One of the simplest systems to get operating is a system which uses only one of the two signals. Briefly, if you have an FSK signal being received by a receiver with either a 500 cycle mechanical filter or a good crystal filter you can tune the receiver so as to pass only one of the two signals. If you then rectify and filter the audio tone coming from the receiver you will have a DC signal that switches on and off depending on whether a mark or a space is being transmitted. This can then be amplified (by relays, tubes, transistors, or magamps) and used to control the printer. It has the viture of simplicity, but unfortunately it also has the accompanying defects. It is quite sensitive to noise and to interference. It also discards half of the information being transmitted, a luxury we can seldom afford with the present level of ham band occupancy. Quite obviously, the transmitting station could save power by merely transmitting either the mark or space only—this is sometimes done and is known as make and break keying (m.a.b.). Some countries do not permit amateur FSK operations and the hams are forced to this less efficient system.

Cycle Counter

Another relatively simple type of filter that is sometimes used is a cycle counter. This circuit, which is used in simple frequency meters, gives a DC voltage output which is proportional to the frequency of the input signal. This voltage output is fed to circuits which give a current on condition for a voltage output below a certain level and a current off condition for a voltage above a certain level. This level is adjustable and is normally set at an output voltage level corresponding to an

input frequency half-way between the mark and space output frequencies. The circuitry is simple, consisting only of resistors, capacitors, and tubes or transistors. No complex adjustment procedures are involved and it is quite tolerant of varying input frequencies and receiver drift. The very tolerance and simplicity which are so appealing are also the qualities which make it less than adequate. It is very sensitive to noise and interference, and while it does a good job when conditions are good, it fails dismally when the going gets rough.

Basic Discriminator

By far the most widely used filter systems are those which employ what is basically a discriminator. The simplest forms are those operating at the intermediate frequency of the receiver and having a peak-to-peak separation of about 1000 cycles. The circuits are substantially the same as conventional FM receivers and, unlike the two systems previously described, have the noise cancellation characteristics of FM discriminators. They can also tolerate a modest amount of drift and will work for signals ranging from relatively short shifts (short shift is the term usually applied to shifts of less than the usual 850 cycles) up to those with shifts of the order of 1000 cycles. This has considerable merit for even the most casual of checks will indicate that while 850 cycles is generally accepted as the standard by the amateur fraternity, some RTTY'ers have only a very foggy notion of what constitutes 850 cycles.

Receiving short shifts is a distinct advantage—you can copy some of the commercial stations using 425 cycle shift, and you can experiment with even smaller shifts. However, if your primary concern is with short shift signals, it would be far better to have a discriminator with a peak-to-peak separation of only 10 percent more than the desired shift. With this circuit the output voltage is proportional to the frequency shift from the center frequency, thus with a 1000 cycle discriminator being used on a 170 cycle shift signal, an interfering signal of equal strength but 500 cycles away from the center frequency will give far greater output voltage than the desired signal.

This technique has another disadvantage however—it will not work conveniently on an AFSK signal. You can, by careful tuning and continuous watching make it operate, but in the process you lose the freedom from drift problem that make AFSK operation on VHF such a pleasant change.

Resonant Circuit Discriminator

The most widely used system is the audio discriminator type which uses one resonant circuit for the mark channel and another resonant circuit for the space channel. With relative low-Q resonant circuits, this is substantially the same as the AF discriminator (see Fig. 13-8). However, it does have the added advantage of being equally useful for both FSK and AFSK. With higher-Q inductors you sacrifice the ability to operate with short shifts; but you improve the interference rejection capabilities of the system, and at the same time decrease the effective bandwidth with a resulting increase in the signal-to-noise ratio. (See Fig. 13-8.) The problem for the amateur RTTY operator has been two-fold, first obtaining inductances with reasonably high Q, and secondly, actually tuning up the unit, once suitable inductors had been procured.

As for inductors, some of the earliest units which were described used conventional chokes or transformers with laminations removed or otherwise modified. These were very low-Q units and while some of them used multiple resonant circuits to improve the performance they were still relatively poor filters. A number of subsequent units used slug-tuned inductors which were designed for television sets. While the Q of these inductances was reasonably good at higher frequencies, their Q and thus their resulting performance at 2125 and 2975 cycles was poor. The best Q and the best performance are to be had from molybdenum permalloy toroids. There are a number of possible sources for these toroids. If you are more affluent than the average ham you can go out and buy them with precisely the inductance you desire for a

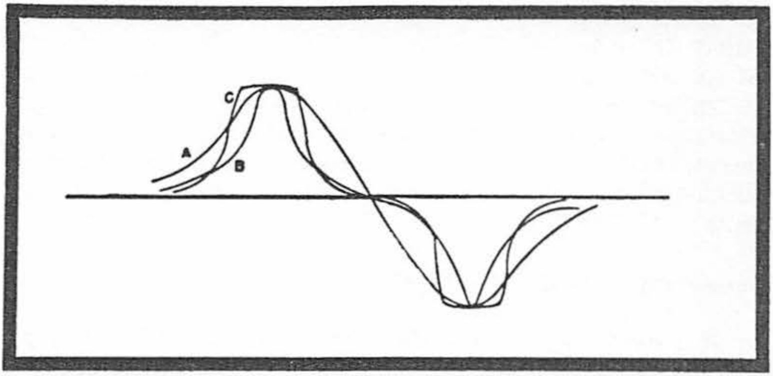


Fig. 13-8. Discriminator response curves. (A) Simple IF type discriminator. (B) Single toroid in each section. (C) Double toroid in each section.

price between $5 and $10 each. A second alternative is to buy the cores for about $1 each and wind them yourself. It is not particularly difficult and you can get precisely the values you want. A third alternative is to find some surplus filters which have usable toroids in them and disassemble the filters. This technique provided about 300 excellent toroids for one group of RTTY'ers at a cost of less than 25c a doughnut. This was the result of a trip to various radio equipment and surplus stores, buying a sample of every likely looking filter, then taking them home and unpotting them to find what was inside. The last and most common source of these toroids is the 88-mh loading coils that are used by telephone companies in large quantities. These are widely advertised and in small quantities cost on the order of $1 each.

Performance

So much for supply problems, back to performance. The 88-mh units used in a single resonant circuit for each of the two frequencies have a number of drawbacks. The response is more sharply peaked than is desirable for general use. Ideally this could be improved by using large inductances which would result in a higher LC ratio—inductances of 500 mh or so are better in single tuned circuits. Another drawback is the fact that using the same value of inductance in each resonant circuit results in a LC that is significantly different in the two circuits (by a factor of about 2 to 1). An added source of variation is the change in Q of the inductors as a function of frequency.

Ideally, you not only want the peak output of each of the two response curves to be equal in value (and this condition is readily achieved by adjusting the inputs of the two circuits), but you also want the area under the two curves to be equal. The latter requirement will result in optimum noise cancellation in the output of the discriminator. This assumes, of course, that by noise you are referring to an input which is stochastically distributed within the relevant frequency limits.

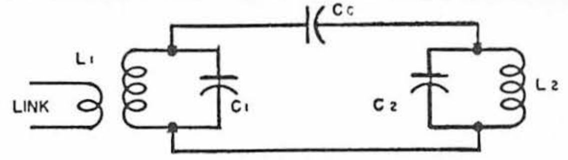
Simple Filter With Two Toroids

The next step toward this ideal and toward improved performance is to use a more complex filter. The most common is a single filter using two toroids in each channel with capacitive coupling between the two resonant circuits. This is just two resonant circuits in which the over-coupling is ad-

justed to broaden the response curves so that the bandpass in both the mark and space channels are the same. This type of filter has been widely used. The basic filter with the appropriate values is shown in Fig. 13-9. The values given are for a bandwidth of 200 cycles. This bandwidth is recommended for general operation, and represents a compromise between a desire for a better signal-to-noise ratio and better rejection of interfering signals that results from a narrower bandwidth—and the ease of tuning, tolerance of moderate drift, and ability to accept inaccurate shifts that result from wider bandwidths. If you have a good stable receiver and don't mind retuning now and then, you can use a narrower bandpass.

Bandpass

The question often arises as to how narrow a bandpass can be used. This will, in fact, depend on the signal-to-noise ratio, the printer speed (baud rate), the characteristics of the filter you use, the number of errors you are willing to tolerate, and the signal processing after the filter. For the average ham, the only two factors he can control in his terminal unit are the filter characteristics and the signal processing after the filter. There is commercial equipment on the market which transmits information at baud rates which are greater than the bandwidth (in cycles) of the circuit, and there is much equipment where this ratio is almost 1:1. Thus, it should be feasible (since amateur operation is at 60 wpm or about 45 bauds) to use filters which are considerably sharper than the 200-cycle figure suggested for general use. The error rate a



	L_1 & L_2	C_1 & C_2	C_c	Link
2125 cps	88 mhy.	.06	.0059	9 turns
2975 cps	88 mhy.	.03	.0022	5 turns

Fig. 13-9. Basic filter with values.

ham can tolerate (or for that matter by quite pleased with), is far higher than would be considered acceptable in normal commercial practice. Luckily languages have a relatively large order of redundancy and an error in one letter of a word now and then does not normally destroy the meaning of a sentence.

Basic Designs

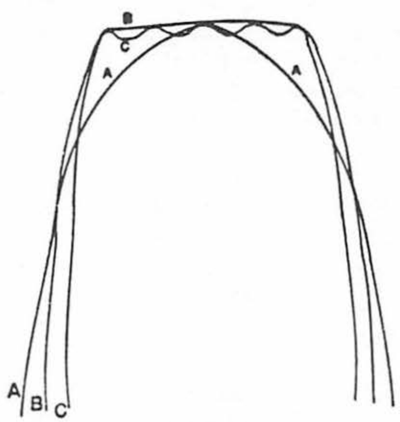
When one starts considering sharp filters another factor becomes important: these filters do not merely have to discriminate between two frequencies, they must respond (and the faster the better), to pulses of the given frequencies which at the fastest reversal rate are 22 milliseconds in length. If the risetime of the filter is too long relative to the pulse length, you will have poor performance regardless of the fine shape of the steady state frequency response curve. You can design three basic types of filters (Fig. 13-10), the most common being the Butterworth or maximally flat amplitude response type. These have a flat response across the top and relatively steep skirts. A second type is the Chebishev, which is becoming increasingly popular. Briefly put, if you are willing to tolerate some ripple across the passband of the filter, you can achieve better skirt selectivity characteristics; and the more ripple you can tolerate the steeper you can make the skirts. The third class of filters are those designed for maximally linear phase variation across the passband of the filter. This type does not have as steep skirts as either of the other types, and it does not have a flat amplitude response across the passband—it is in fact peaked in the center of the passband and falls off gradually toward the edges of the passband. It does have a significant advantage, it has a faster risetime than the other types of filters and hence is more appropriate for use with pulse inputs. It is also less subject to ringing which often becomes a significant problem with sharp, steep sided filters. If you want to try better filters than the simple two section variety described earlier, by all means use the maximally linear phase characteristic.

Since the objective is to recover from the two tones information in the form of 22-millisecond pulses, the fastest possible case (successive mark-space signal reversals) is a square wave of about 23 cps. You must recover the third order products in order to reasonably reconstitute this square wave which was being transmitted, if you have no provision for pulse reshaping following the discriminator.

Square Wave Input

Each audio tone may be considered as consisting of a carrier on the particular center frequency (2125 or 2975) which is modulated by a square wave. At the fastest reversal rate you have 22 millisecond pulses or a square wave of approximately 23 cycles, this is approximately what you have when RYRYRY...is being transmitted. The average modulation rate is lower than this, but this represents the maximum requirement of the system. This square wave produces the usual Fourrier series of sidebands, thus you have the 2125 cycle carrier, the first order sidebands at 2125 plus or minus 23 cycles, the third order sidebands at 2125 plus or minus 3x23 cycles, and all the rest of the odd order sidebands. The major portion of the energy is in the carrier and the first order sidebands, with decreasing amounts of energy in the higher order sidebands.

We can separate the input to the filter into three basic components: the tone carrier, noise (by which we mean



Filter response curves:
a. Maximally linear phase design
b. Butterworth (maximally flat amplitude)
c. Chebyshev

Fig. 13-10. Filter response curves. (A) Maximally linear phase design. (B) Butterworth (maximally flat amplitude). (C) Chebishev.

stochastically distributed inputs), and frequency (by which we mean signals with a coherent frequency distribution which may appear within the passband of the filter). When we narrow the bandpass of the filter we are affecting all three inputs: 1. We are discarding higher order sidebands of the desired signal; 2. We are decreasing the noise, and 3. We are decreasing the probability of receiving an interfering signal. The first mentioned effect is the price we pay for the advantages of the last two. When we go from a filter that passes the fifth order sidebands to one that passes only the third order sidebands we find we have decreased the noise to 60 percent of the former value and the probability of interference by the same amount, and since the loss in information in the fifth order sidebands is not this large we increase the signal-to-noise ratio and the efficiency of the system. Now if we decrease the filter from a bandwidth that passes the third order sidebands to that which passes only the first order sidebands we have decreased the noise by a factor of three, the probability of interference by the same amount, and the information we have lost is substantially less than this amount. Don't try to take it any further or you start losing ground. Note that the modulation rate is not normally the 23 cycle square wave, it actually varies from nearly zero to this figure. This is not to imply that you cannot go further—there is a theorem in information theory that says you can transmit an infinite amount of information in an infinitely small bandwidth if you have an infinitely high signal to noise ratio. The last is what you don't have. Even to go to the limit indicated of a 46 cps bandpass which recovers only the first order sidebands would require perfect stability, excellent filters, and the best in signal reprocessing techniques.

Square Wave Restoration

Any terminal unit of reasonable design will, in fact, incorporate some facility for pulse reshaping—perhaps the most common and elementary device is a relay. With inputs above a certain level this will be on and at inputs below a certain level it will be off, and if you feed low frequency sine wave input to it you will get a square wave out of it. There is, in fact, no requirement for a reasonably square wave input. It has the obvious disadvantage that the hystersis or difference between pull in and drop out levels is too great and that the adjustments of the characteristics are usually mechanical and extremely difficult. Because it is a mechanical conversion device the speed at which it reacts is limited. Further it tends to stray

from optimum adjustment over time and has contacts which generate a species of noise which is very difficult to eliminate.

The same object can be accomplished electronically, and it can be done in a fashion so that the action can be readily adjusted and the noise of mechanical contacts eliminated. Probably the simplest electronic means of reconstituting a square wave is to overdrive an amplifier that is fed by the degraded waveform. This technique has been widely used and while it gives relatively good square wave output it lacks flexibility, and is it possible with certain settings to get intermediate outputs which are neither on nor off. A better technique—which can be easily adjusted over a wide range and gives only on and off outputs—is the Schmitt trigger. These can be readily built with either tubes or transistors (although with this circuit the transistor version is simpler and easier to work with).

Schmitt Trigger

Fig. 13-11 gives an excellent illustration of the action of a Schmitt trigger when the trigger level is varied over the

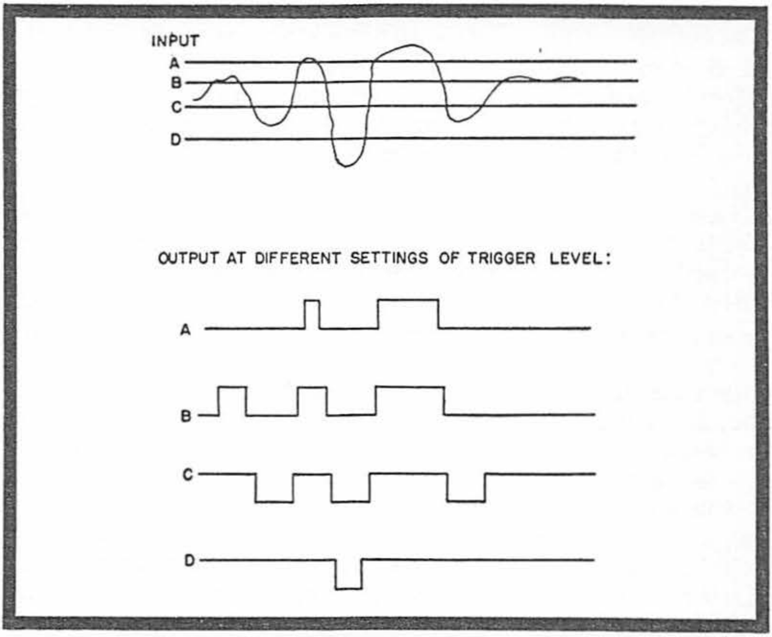


Fig. 13-11. Action of Schmitt trigger with irratic input levels.

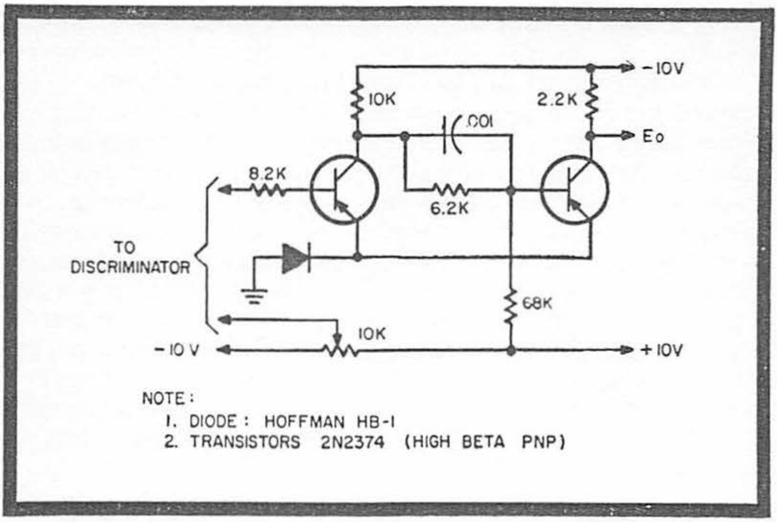


Fig. 13-12. Schematic of typical Schmitt trigger using transistors.

operating range. Here the positive direction corresponds to a mark signal and the negative direction to a space. Levels A, B, and D give misprints, while setting C gives a practically perfect letter F. Note here that regardless of the input voltage wave shape that the output from the trigger is a perfect square wave. The filters do not have to deliver a reasonably square output—they merely have to separate the input signal into two channels and deliver the resulting outputs to the discriminator. The Schmitt trigger then electronically makes the decision as to whether a mark or a space is being transmitted. This decision is based on the relative amplitudes of the outputs of the two channels. The bias level setting compensates for continuous unwanted signals in either channel, or in the case of AFSK for variations in the levels of mark and space tones being transmitted. (This varies more widely than you would suspect, having these within 25 percent is better than par for the course.)

Fig. 13-12 gives the circuit of a typical transistor Schmitt trigger. The trigger level without the bias potentiometer is about -.8 volts, but with the variable bias this point may be varied over a plus or minus 10 volt range. Provision should be made in the terminal unit to insure that the discriminator output voltage range does not exceed this 20 volt peak-to-peak range. In actual practice there is a small amount of hystersis in the circuit, the output flips to the on condition when the input

exceeds -.83 volts and turns off when the input falls below -.80 volts. This difference of .03 volts is of no import when compared with the 20-volt range of the input from the discriminator. Compare this with the typical relay which has a 2:1 range between pull-in and fall-out.

It should be noted that even if no explicit restoration technique is used in the terminal unit, you will get it in the printer as a result of the action of the armature of the selector magnet of the machine. This is the poorest of all techniques, adjustment will be critical, and the range of the machine will be very narrow. This function should be provided in the terminal unit, should have minimum hysteresis, and should have as wide a range of adjustment as possible.

Filter Tuning

The next step, once you decide on the filter you want, would logically be to tune up the filter. But for those who don't have counters available it might be best to turn your attention to some sort of frequency standard. If tuning up the filter is going to be a one shot affair, the most reasonable procedure is to use the standard audio tones broadcast by WWV or one of the other standard frequency stations, along with an oscilloscope to give you the appropriate Lissajous patterns. After your audio signal generator has warmed up, completely calibrate it as accurately as possible in the 2000-2250 and 2850-3100 cycle regions. For those who desire greater precision than this, or plan to do an appreciable amount of work, either borrow or build a standard for the common RTTY frequencies. There are two types of standards in use, both of which do an admirable job. The first and most common—although not the easiest to build is the tuning fork standard. This uses a tuning fork as a resonant element and normally operates on a frequency of 425 cycles. These are made with standard 435 cycle tuning forks which have been lowered to 425 cycles either by loading the ends of the tynes with solder or by filling the crotch of the fork. They serve admirably for tuning up filters and checking RTTY equipment. With an oscilloscope they give you 5:1 and 7:1 Lissajous patterns with the standard 2125 and 2975 frequencies. They are also handy f r setting the 850-cycle shift in FSK operation. The alternative approach is to build a crystal oscillator for 446.25 kHz (there are standard surplus FT-241 crystal units that are approximately this frequency. They may be easily brought to precise frequency by edge grinding the crystal). The oscillator is followed by two successive multivibrator dividers—the first dividing by 6 and the second dividing by 5. The end result is a standard

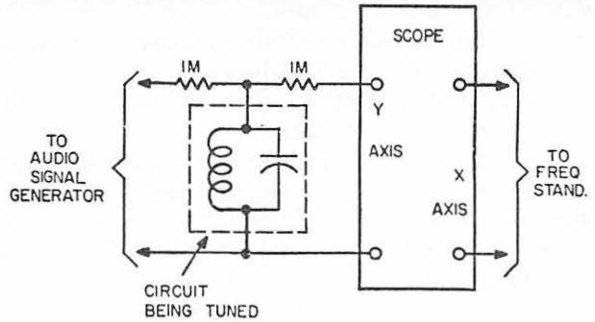


Fig. 13-13. Setup for tuning a filter circuit.

frequency of 14,875 cycles. This, too, gives the 5:1 and 7:1 Lissajous patterns with the standard frequencies. If you intend to do much RTTY work, they are handy gadgets to have.

The next requirement is a voltage indicator, if you are using a counter or a well calibrated audio signal generator, a good AC VTVM is all that is needed. If you are using one of the frequency standards, an oscilloscope is highly desirable since you can check amplitude and frequency at the same time. The basic setup is to feed the output of the audio signal generator through an isolating resistor to the circuit under test, and in turn, through another isolating resistor feed the Y axis of the oscilloscope. (See Fig. 13-13.) The two resistors should be 1 megohm or higher. The standard signal generator should be fed to the X axis of the scope to permit observation of the Lissajous patterns for calibration purposes. With this set up you tune the circuit for resonance (which is indicated by maximum amplitude) at the proper frequency (which is indicated by the Lissajous patterns).

Tuning the circuits to resonance is naturally the basic process of varying L or C. With the usual 88-mh toroids there is usually not much variation from one unit to the next. They seem to be wound to relatively tight tolerances. The easiest method by far is to have a large number of capacitors of approximately the correct value, and just keep trying different ones until you find the right value. The typical spread on capacitors is fairly wide and the selection process is fairly simple. A similar technique which requires fewer capacitors is to use two capacitors to get a given value—the .06 mfd that is needed to tune up the 2125 Hz unit may be obtained from an .01 and an .06 in parallel, and resonance can be achieved from a

relatively modest supply of each by successively trying various combinations of the two. Of course, if a given capacitor falls slightly below the desired capacitance, a small capacitor can be added for precise resonance. If your stock of capacitors is small, the easiest adjustment is in the number of turns on the toroid. It is easy to add a few turns or peel off a few to tune the circuit to resonance.

A simple two-toroid filter and approximate values were given in Fig. 13-9. There are two basic ways to attack this problem. The first, and most elegant, is to tune both L1-C1 and L2-C2 to a frequency of 100 Hz (i.e., half the desired bandwidth of the filter) higher than the desired center frequency. In this case the frequencies would be 2225 Hz for the pair intended for the mark filter and 3075 Hz for the pair intended for the space filter. If you have a 425 fork standard you will find it quite convenient to use a 21:4 Lissajous pattern, which gives you a frequency 425-4 or 106 Hz high. On the space filter use a 29:4 Lissajous. Then assemble the filter and try successive values of coupling capacitors—the correct value will extend the passband of the filter to the desired limit on the lower side of the center frequency. The larger the value of coupling capacitor the wider the bandpass of the filter. For narrower filters, the pairs are initially tuned to frequencies closer to the center frequency and the coupling capacitors are not as large. An excellent approximation of the coupling capacitor can be obtained by shunting a single LC circuit with the coupling capacitor. The circuit should then tune to the center frequency.

The other method is to select the value of the coupling capacitor (Cc in the table) and tune each pair individually to the center frequency. Thus, you will first resonate L1(C1 + Cc) to 2125 Hz, then remove Cc from this circuit and resonant L2 (C1 + Cc) to 2125 Hz, and finally assemble the filter in the final circuit configuration. This is easier than the first method and only requires the use of the 2125 Hz reference frequency rather than shuttling back and forth between the two sides of the center frequency. This procedure should best be followed with the filter in place or at least with the operating load circuit connected since when it is actually installed there will normally be added capacitance across at least the last of the two toroids. Merely shorting out the toroid not being tuned will place the coupling capacitor across the appropriate resonant circuit.

You will quite probably find when you have finished the tuning that the passband is not flat. This will depend on the loading of the filter. The response can be smoothed out by

adding a resistor across the final toroid. Do not do this until it is installed in the operating circuit, however, since this may provide sufficient loading. If you still have a double-humped response add a resistance of whatever value is required to smooth out the response. This value of resistance will not be the same for both the mark and the space filters, since the loads will be approximately the same for the two while their impedances vary by a factor of about 2:1.

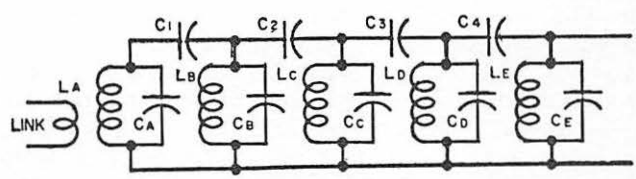
The values given are for the standard 88-mh units, but if you have other toroids you can merely adjust the capacitance values by the appropriate ratios and use the same tuning procedures. If you have different inductance toroids, you should use the larger inductances in the 2125 Hz filter and the smaller inductances in the 2975 Hz filter.

The input to the filter is easily accomplished by winding turns on the toroids. The output section preceding the filter should use a plate to voice coil or collector to voice coil transformer. The voice coil winding provides matching to the links wound on the toroids. The number of turns on the links are adjusted to give equal output from each of the two filters. This adjustment, while not difficult, is important since the outputs of the two filters are so arranged that when there is an interfering signal or noise with equal amplitude in both channels, the resultant outputs cancel. We found 9 turns on the 2125 filter and 5 turns on the 2975 filter gave these desired results, but you should check the performance before you consider the job complete.

Multi-Section Filters

The next step in order of effectiveness, and of complexity too, is the use of multisection, maximally linear phase filters. Getting set up to tune filters is the major chore, and the actual tuning is a relatively straight forward and not particularly time consuming task. These filters may be designed for as many sections as you wish, and for whatever bandwidth is desired. One using five toroids in each channel and the basic circuit is shown in Fig. 13-14. The bandwidth chosen, 85 Hz, is a compromise between the requirements for optimum signal-to-noise ratio plus interference rejection capabilities and the necessity of accommodating a moderate order of drift plus not having hypercritical tuning requirements. The performance is excellent, particularly when the going gets rough. The skirt selectivity is impressive indeed and, in practice, the tuning is not particularly difficult. The tune up procedure is the same as that outlined in the latter part of the discussion of the

procedure for two section filters. The values of C1, C2, C3, C4 are not critical and are either ordinary 10 percent tolerance units or combinations of them. Mount and wire up the filter with the toroids and coupling capacitors in place. For a convenient mounting, ordinary terminal boards serve nicely with the terminal points providing mountings for the capacitors and the toroids mounted on alternate sides of the board. One note of caution on mounting toroids in general: don't have them completely surrounded by metal—it is equivalent to a shorted turn and the performance is very poor. To tune up the first section, short the second section and tune the first using the same setup described before—either by varying the capacitance or adjusting the turns on the toroid. When this is completed remove the short from the second section, short the first and the third section and proceed to tune up the second section. The successive sections are then tuned up in precisely the same manner, always making sure that the short has been removed from the section you are tuning and that the two adjacent sections are shorted. For the final section, which, when the filter is operating will be connected to additional load, is beat tuned after the unit is connected to the terminal unit. The coupling is accomplished in precisely the same was as before. You can try the same number of turns suggested before as a starting point. Although, the values given here are for 88-mh toroids, other values can be used by making the proportional changes in capacitance. This assumes that the Q of the toroids you use is not too far off from those of the 88-mh units; the design of the



(Approx.)

freq.	C_1	C_2	C_3	C_4	C_{a-e}
2125	.0074	.0037	.0026	.002	.06
2975	.0025	.00125	.0009	.0007	.03

Fig. 13-14. Values for 85-Hz bandwidth, maximally linear phase response, using 88-mh toroids.

linear phase shift filters is based on the Q of the toroids as well as the frequency and bandwidth of the filters.

A further point to note is implicit in the tune up procedure. Ca has C1 in parallel for resonance, Cb has C1 and C2 in parallel, and so on—hence the values of capacitance will not be the same for Ca through Ce. The actual values may readily be computed since you need .064 mfd to resonate 88-mh at 2125 Hz and .033 mfd to resonate 88-mh at 2975 Hz. Thus the actual values will run under the approximate values given. This has the merit of using different values for each capacitor—the probability of your having a quantity of identical value capacitors is relatively small. Some mention should also be made for those few who will actually consult the references given on these filters. The standard design lists a resistor across La to degrade the Q of this section for optimum phase performance. In actual practice the loading by the input circuit takes care of this problem handily, but if there is any doubt as to whether you have enough turns on the input link—by all means err on the side of overcoupling.

Filter Construction

The filters were made on the terminal boards mentioned before, and after the filters were tuned, they were both mounted beneath a 5" x 10" x 3" chassis with 4 leads, 10 inches long coming from each filter to an octal plug. By removing the regular two-section filters (which were in Vector C-12 cans with octal bases) and plugging in the two octal plugs we are able to convert from the broad to the sharp filters in a matter of seconds. Since our terminal unit is built on a 5" x 10" x 3" chassis, also, we merely place the TU on top of the filter and have a very compact unit. (See Fig. 13-15.)

You will find the performance of this filter decidedly better than the simple two toroid filter. The passband is adequate for good performance and the unit has high selectivity without the attendant ringing of other designs. If you see the conventional scope tuning indicator with the output of the mark channel to the horizontal plates of the scope, and the space channel to the vertical plates of the scope, you will note that the lines are considerably sharper with this filter. In addition, since the passband is not flat but is slightly peaked in the center (the 85 Hz bandwidth figure here specifies the width to the -3 db points) the tuning is actually easier in some respects, since there is no ambiguity in the setting. However, don't throw away your old filters, there will be times when you need them—particularly when the other station is using only a

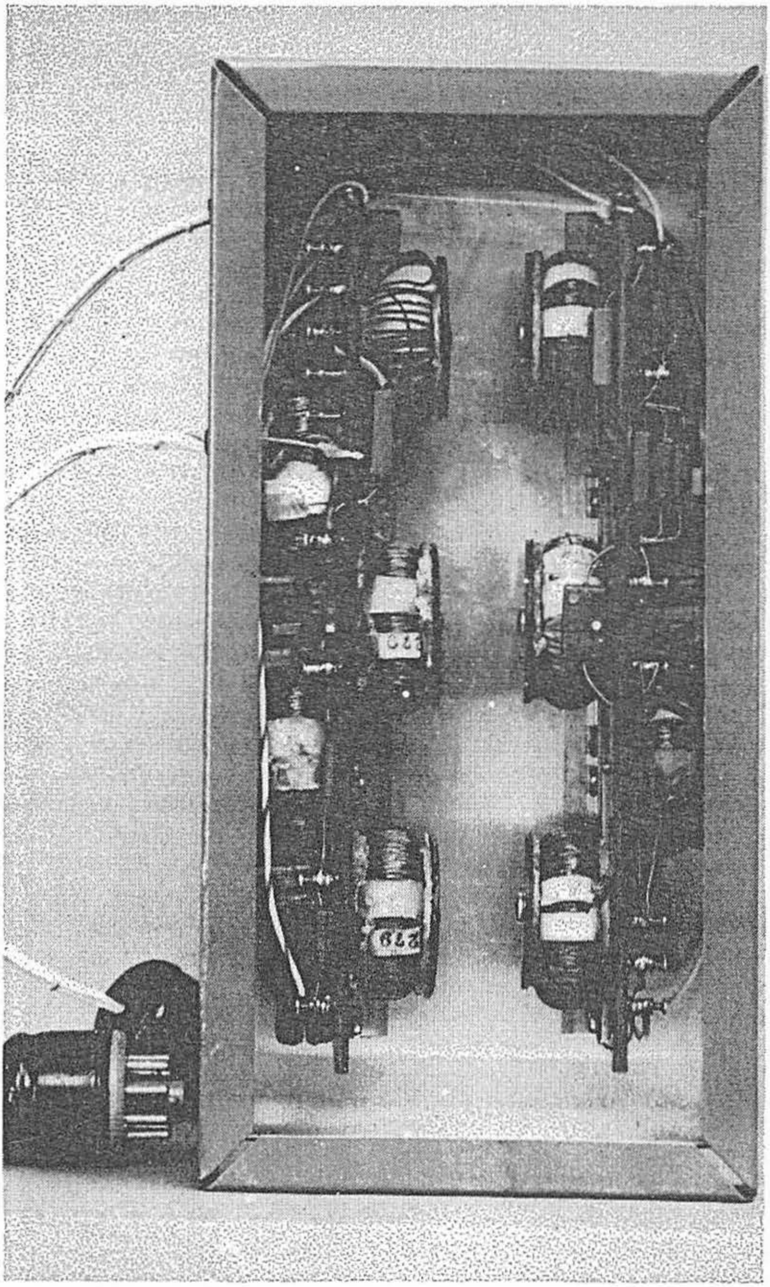
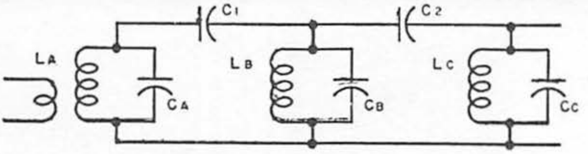
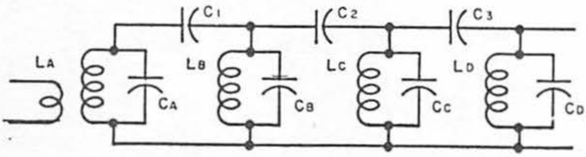


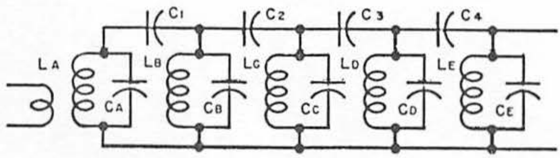
Fig. 13-15. A five-section maximally linear phase filter. This does not use 88-mh coils, but the techniques are identical and the LC ratios are appropriately scaled.



Width	Freq.	C_1	C_2	C_{a-c}
60	2125	.0027	.0014	.06
60	2975	.00082	.00048	.03
85	2125	.0043	.0021	.06
85	2975	.0014	.0008	.03
120	2125	.0062	.003	.06
120	2975	.0021	.0011	.03



Width	Freq.	C_1	C_2	C_3	C_{a-d}
60	2125	.0038	.0020	.0014	.06
60	2975	.0011	.00064	.0005	.03
85	2125	.0058	.0029	.0020	.06
85	2975	.00195	.0010	.0007	.03
120	2125	.0087	.0042	.0029	.06
120	2975	.0029	.0015	.00105	.03



Width	Freq.	C_1	C_2	C_3	C_4	C_{a-e}
60	2125	.0049	.0024	.0018	.0014	.06
60	2975	.0015	.0008	.0006	.0005	.03
85	2125	.0074	.0037	.0026	.002	.06
85	2975	.0025	.00125	.0009	.0007	.03
120	2125	.0113	.0055	.0038	.0029	.06
120	2975	.0039	.0019	.0014	.0010	.03

Fig. 13-16. Design data for 3-, 4-, and 5-section maximally linear phase shift filters, providing a choice of bandwidths.

rough approximation of 850 cycle shift. The ideal arrangement would be one in which you could merely switch from the two section to the five section filter. However, in actual practice we have found the plug-in technique quite adequate.

The most significant advantage in short shift is a result of the propagation characteristics of electromagnetic waves; the closer two frequencies are to one another, the small the probability of selective fading. Since the basic method of detecting the RTTY information is to make a decision based on the relative signal in the two channels, it is desirable to have a minimum amount of selective fading. Another advantage for many is the possibility of supplementing the selectivity of the terminal unit with the more selective IF filters which are found in the better ham receivers. A further advantage is the possibility, with the more compact signal that results, of avoiding certain types of interference.

Plug-In and Switching Changeover

An advantage of the plug-in technique used for the filters we have mentioned is that you can build up one extra section for short shifts. If you have filters for 2125 and 2975 and are interested in short shift, say 170 Hz, you have only to build a unit for 2295 Hz and use it in conjunction with the 2125 Hz unit for this short shift work. In addition, you may also want to build a unit for 425 Hz shift for copying a number of the commercial stations using that shift.

The first filter is the hard one—once you have gone through the procedure, the rest are easy and the returns in terms of actual on-the-air performance are impressive. If more attention is turned to putting together a satisfactory terminal unit with adequate filters you will get much more effective and much more enjoyable RTTY operation.

For the adventurous—and for those who don't like graphs, mathematics, and such, Fig. 13-16 is included. This gives design data for 3-, 4-, and 5-section maximally linear phase shift filters, with a choice of bandwidths which you can use according to your receiver stability and personal tastes and inclinations. One of the reasons both the maximally linear phase shift filters and the Chebishev filters were not more widely used before was the mathematics involved. More recently tables have been compiled which make the procedures far easier. The advent of electronic computers that make formerly impossible tasks merely the subject of a modest amount of machine time have changed these from the realm of interesting theoretical possibilities or the objects of arduous cut and try procedures into relatively common everyday items.

CHAPTER 14

RTTY HANDBOOK

Autostart

The VHF amateur who is interested in establishing dependable communications, between his friends in his own area should be interested in a method of improving that dependability. It is called autostart Teletype and is used in amateur activities that center around one particular channel which is used by many people as a calling frequency. Although employed to a limited extent on the low bands, autostart Teletype proves to be most useful between club or net members using the VHF bands for local communications.

Autostart means automatic starting. As an example, suppose you have a receiver monitoring your favorite channel. While you are out, one of your friends calls you on the radio to give you a message. When you do not answer, he turns on his teletype tone. Your Teletype machine starts up, prints his message and shuts itself off again, all automatically. When you come home the message is there, waiting to be read. The possibilities of this system are virtually limitless.

TELETYPE DECODER

The Teletype decoder (converter) presented here was designed for VHF activity using AFSK. It may also be used for HF frequency shift RTTY by leaving out the connections to the autostart circuitry: A7, Q8, Q9 and the autostart relay. It was revised a few times to make it simpler, cheaper and more compatible with existing equipment. The result is a solid-state unit which will run for years in continuous use without maintenance. It is simple to build and will cost very little in the way of parts. The printed circuit board for the unit is larger than necessary so as to provide plenty of space for odd-sized parts. It may be copied directly or you may wish to design your own.

The decoder works in the following manner: The input audio is limited, amplified and then split to separate channel filters. Q2 is the 2125 Hz (mark) filter and Q3 is the 2975 Hz (space) filter. The audio is then rectified and filtered to DC

which is fed to a divider network consisting of two 56K resistors. With respect to ground, the junction of these two resistors swings positive and negative on mark and space tones.

Transistors Q4, Q5 and Q6 form the DC coupled amplifier which drives the printer magnet and turns the current on and off. The diode D7 helps shut off Q6 completely and also acts as a fuse to protect the transistor junction against excessive currents.

Transistors Q7, Q8 and Q9 also act as DC switches. The zener diode D8 prevents triggering on random noise and voice. The resistor-capacitor network between Q8 and Q9 forms the on-off time constants for the autostart section. There is a delay of about one second from the time the mark tone is applied to the input until the relay pulls in and about three seconds delay for the relay to drop out when the tone is released. These time constants prevent the machine from coming on during noise and voice reception and also prevent the machine from shutting off during high space content transmissions.

The relay employed at the output of the autostart section is a small, sensitive, 24 to 30-volt sealed type which only requires a few milliamperes of current to pull in. If you can't find this type, a higher current relay can be used provided that Q9 is chosen so as to handle the power. The contacts of any relay you use must be capable of handling 110 volts AC at about 2 amps. This means that it will be necessary to drive a large 24-volt relay from the small one, if it is used.

Many different types of transistors will work in this unit. Try the ones you have before you go out and buy. Except for Q6 and Q9 the parameters are not critical. The types shown in Fig. 14-1 are typical. General purpose, silicon audio types with medium betas will serve just fine. For Q6 a power transistor in a TO-3 case may be used. It will require no heat sink. Select one with a low leakage current. For Q9 be sure it will handle the relay current you plan on using. The one specified in the schematic is good for about 50 milliamperes.

Once the components are mounted, the decoder can be tested in the following manner. Check all diodes to be sure they are in correctly. Remove Q4 and the zener diode D8 from the circuit and apply power. Place an audio tone of either mark or space on the input and check for audio with a scope or high impedance crystal earphone at test point 1. At point 2 the level should be higher. At point 3 the mark tone should be stronger than the space tone and at point 4 the space tone should be stronger than the mark tone.

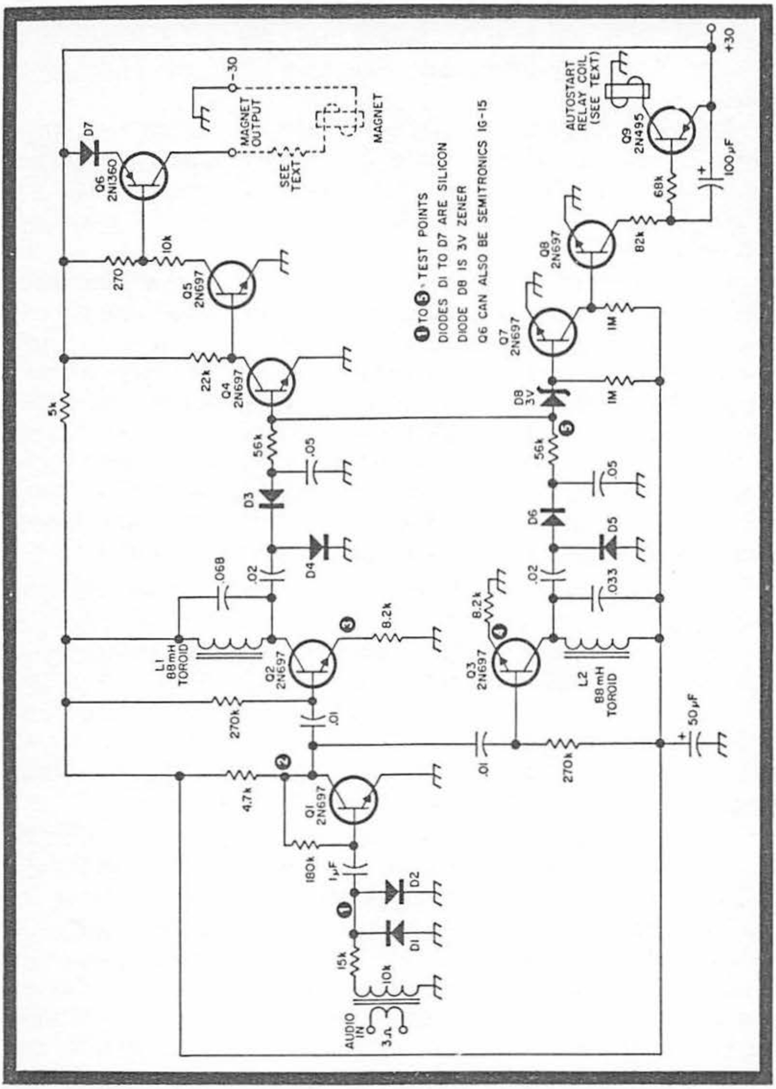


Fig. 14-1. Schematic for decoder (converter) for VHF AFSK autostart RTTY. It can also be used for HF RTTY, but the autostart circuitry may prove unreliable for this use.

Connect a VTVM to test point 5 and switch back and forth between mark and space tone. On mark the voltage should go about 5 volts negative and on space it should go about 5 volts positive (Q4 and D8 must be disconnected for these readings).

The swing can be greater, but less than plus or minus 5 volts indicates possible malfunction. If the mark and space voltage at point 5 differ by more than .3 volts it would be advantageous to adjust R1 and R2 so the gain through both channels is equal. This point is a discriminator type output which, like an FM receiver, will tend to cancel noise and random audio interference.

If all checks out, insert Q4 in the circuit and adjust R12 until the magnet current (Fig. 14-2) is 60 milliamperes with the mark tone on. This will allow either series or parallel configuration on the magnets by simply adding a resistor in series with the magnet for 20 milliampere operation. The value shown is a nominal value. Switch to space tone and the magnet should no longer hold in, even when manually depressed.

Finally, insert the zener diode D8 in the circuit and apply the mark tone to the input. After approximately one second delay the autostart relay should pull in. Release the mark tone and measure the time it takes for the relay to drop out. This should be about three seconds. These times can be longer, but will cause trouble if they are shorter.

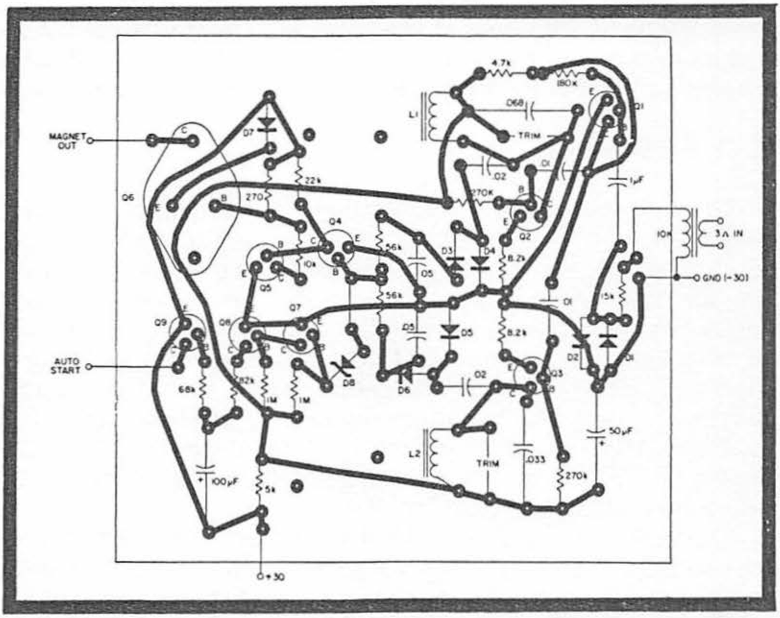


Fig. 14-2. Copper layout for the etched circuit AFSK autostart decoder (converter). Components on opposite side of board are shown by lighter lines.

This completes the checkout of the Teletype-decoder and it should be ready to place in service. The range of the machine will be slightly narrower than on local loop but if the decoder is adjusted properly the loss should not exceed 10 points.

TELETYPE ENCODER

The Teletype encoder is really an audio oscillator with a transistor switch and is used for the transmission of Teletype signals by modulating the carrier. Two tones are used, the mark tone (2125 Hz) and the space tone (2975 Hz).

The encoder can be built on the printed circuit card (Fig. 14-3) or any other way you may choose. Only frequency determining components are critical in this circuit and many types of transistors will work equally well here. The audio output is enough to drive any transmitter including carbon mike types. An external potentiometer should be used to control the level of the generator.

The only difficulty you may encounter is getting Q4 (Fig. 14-4) to switch from space to mark. First, try a few different types of transistors and if that fails juggle the values of R10 and R12 around. The beta of the transistor is most likely the gremlin in the problem. If it is too low the mark tone will not come on and if it is too high the space tone may not come on.

RTTY AUTOSTART

This autostart circuit is versatile in that it connects to the speaker instead of digging into the TU, is solid state and therefore small, and can be hooked up to turn on the teletype machine and TU power (Fig. 14-5). The power supply requirement is +24 to +28 volts DC at 10 ma no signal to 200 ma with signal. This autostart was designed originally for monitoring USAF MARS VHF nets, where both voice and teletype are used. Many stations operate on VHF and HF SSB simultaneously. With this device it is not necessary to stop copying on HF to turn on the teletype machine. It insures that you receive all RTTY copy on the net.

The first transistor is a tuned amplifier. The input audio must be in the range of 1 to 5 volts peak-to-peak which is the normal listening range voltage for a 4 to 8 ohm speaker. For a 600-ohm impedance, a resistive divider consisting of a 1K and 10K resistor or a 600- to 8-ohm transformer must be added. The tuned circuit is resonant to the standard mark frequency of 2125 Hz. Any combination of L and C may be used for other

tone frequencies for selective call. Bandwidth is approximately 150 Hz. The .068 mfd capacitor should be a good mylar type for frequency stability. The coil is the usual 88-mh toroid available on surplus. Turns may have to be taken off to get exact resonance for best performance. If the .068 capacitor is close to tolerance, about 35 turns should be taken off. The link couples about one volt at resonance to the switch transistor Q2 which then turns on and allows capacitors C3 and C4 to charge up. When this charging voltage gets up to 1½ volts the relay drivers Q3 and Q4 turn on and energize the relay. The 100K resistor with a 100 ohm relay gives about 3 seconds before turn on. Likewise, it takes about 3 seconds to drop out in absence of a steady mark tone. The time out is determined by the relay coil resistance and the 22K resistor. Both the 100K and 22K resistors can be varied for any desired time in and out, up to 10 seconds.

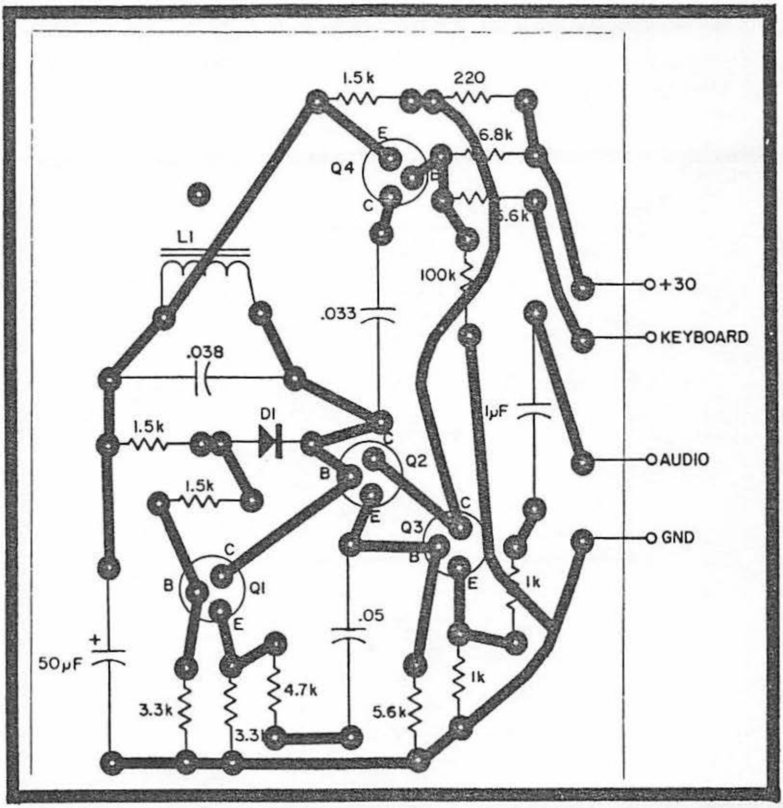


Fig. 14-3. Copper layout for AFSK RTTY encoder (generator). The components on the opposite side of board are shown by lighter lines.

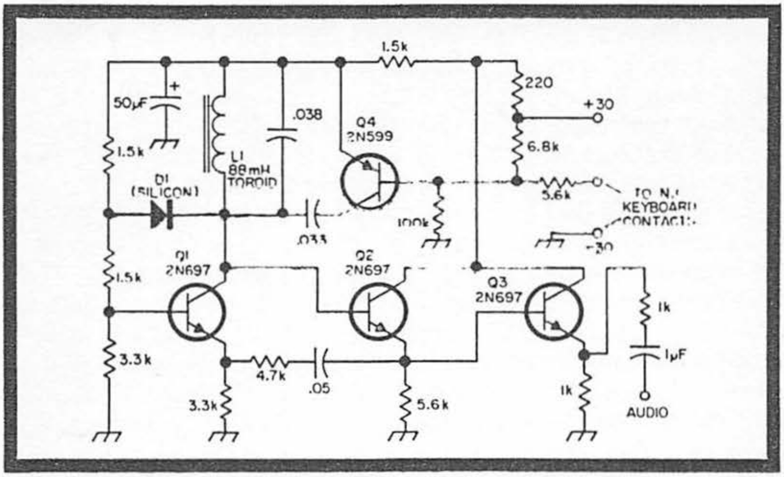


Fig. 14-4. Schematic of AFSK encoder (generator). This circuit generates 2125- and 2975-Hz tones used to modulate AM transmitters for VHF RTTY.

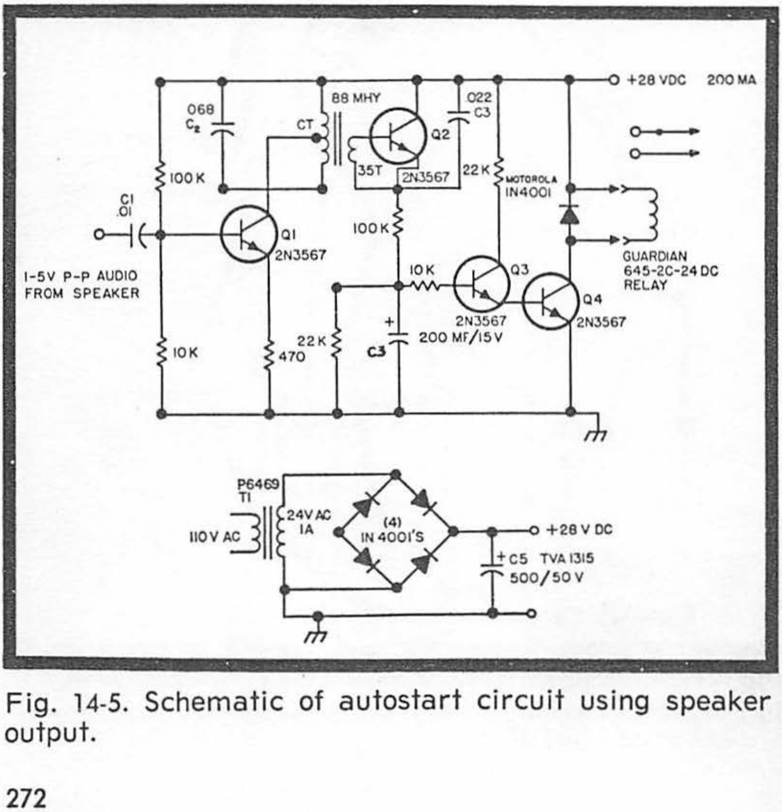


Fig. 14-5. Schematic of autostart circuit using speaker output.

Operation

Hook up the input to an audio source of 2125 Hz and take off 5 turns at a time from the hot side of the toroid until maximum voltage is shown on a scope or AC VTVM. Check the time in and out of the relay and change the 100K and 22K resistors for the desired length of time. The relay should have 10 ampere contacts because of the starting current of the teletype machine.

SELCAL

The Selcal (Fig.14-6) is sort of an electronic stunt box. It receives RTTY characters directly from the loop, with no machinery running. It recognizes four (or more) characters, in the proper sequence. An output relay closes to turn on your printer or other device. It then recognizes receipt of four letters N, sent at the message end, to turn off your printer. While the characters must be received in the proper sequence, the Selcal does not distinguish between upper and lower case. Fig. 14-7 shows how the Selcal is hooked up.

The basic system is very versatile, and will be the basis of further RTTY logic systems such as regeneration, series-to-parallel conversion, and speed conversion.

The system is digital, using inexpensive Motorola integrated circuit (IC) logic blocks. This logic is designed to operate in practically any combination, with voltages, switching times, etc., figured out for you, eliminating much circuitry detail. Best of all, they work! Their cost is far below even junk box prices.

Logic

The Selcal is built entirely of three types of logic. This logic series operates on two voltage states: high (H) (over 0.8 volts) will turn on any gate, low (L) (under .43 volts) ensure all gates are off. Levels between .43 and 0.8 volts would give erratic operation and are not used. The logic symbols do not show the B+ (3.6v) or ground connections.

Inverters

The simplest type of logic is the inverter, shown in Fig. 14-8. This is just a resistance-coupled amplifier designed so that in the on state the output is less than 0.43 volts. The inverter has a small "logic gain," or fanout, meaning one stage will drive several succeeding stages. A buffer is similar to an

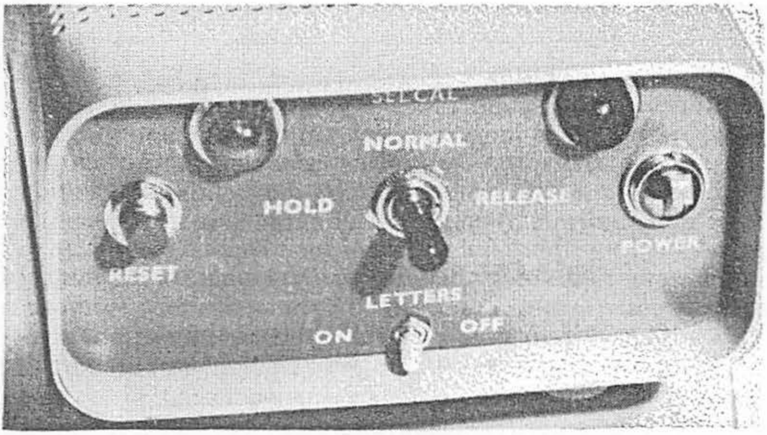


Fig. 14-6. Selcal (Selective Call) unit.

inverter but has a greater fanout capacity, and is available in both inverting and noninverting circuits. The MC789P Hex Inverter contains six independent inverter stages.

NOR Gate

The next logic type used is the NOR gate, shown in Fig. 14-9. It is obvious that if any input is high, a transistor will be saturated and the common output will be low. Only if all inputs are low can the output be high. The NOR gate is a most universal function, and nearly all digital computer circuits and systems can be built from combinations of this logic type. In the Selcal, we will use the NOR gate as a coincidence recognizer. With varying high and low signals on all inputs, there will be an output only at the instant all are low.

Flip-Flops

The J-K is an unusual but most versatile type of FF used in modern digital systems. It is also called a master-slave, or clocked flip-flop. Its symbol and operation table are shown in Fig. 14-10. The inputs are: Set (S) and Clear (C) (sometimes called the J and K inputs), toggle or trigger (T) and preset (P). The outputs are (1) and (0), sometimes designated as (Q) and ($\overline{Q}$). These outputs are always in opposite logic states; that is, when one is high the other is low. The preset function is not shown in the truth table. When the (P) lead is high, the (1) output is forced low, regardless of the states of the other inputs. The integrated-circuit J-K contains the equivalent of 15

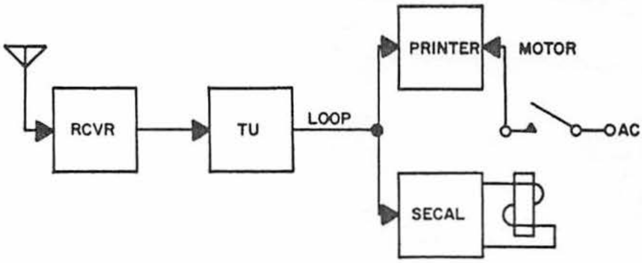


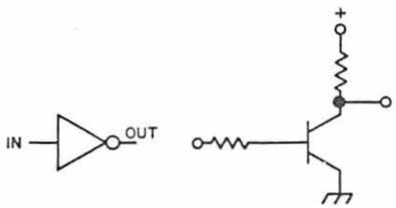
Fig. 14-7. Connecting the Selcal into a RTTY system to turn on the printer when the proper call letters are received.

transistors. Two independent circuits are contained in the Motorola MC790P.

The J-K can be connected for several different logic functions. Fig. 14-11A shows the J-K uses as a common binary counter, or divide-by-two circuit. This divider will be used to count down the oscillator frequency in the Selcal. Fig. 14-11B shows it as a set-reset flip-flop. Fig. 14-11C shows the clocked flip-flop operation. For this use the (S) and (C) inputs must be in opposite states, so an inverter is used as shown. The output logic states duplicate the input states after the clock pulse. This FF is seen to be timed, or clocked. It will be used in this mode in the Selcal Shift Register. The truth table in Fig. 14-10 shows all modes of operation.

Basic Operation

The Selcal is basically a series-to-parallel converter. The five character-information pulses, mark or space, are briefly



IN	OUT
H	L
L	H

Fig. 14-8. Inverter logic.

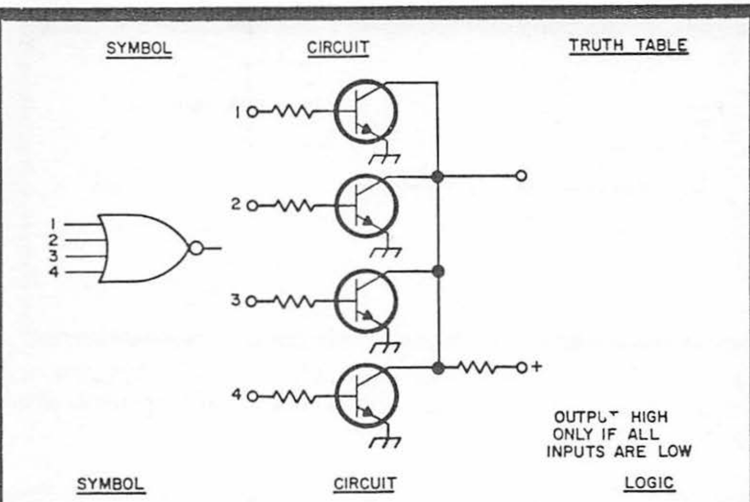
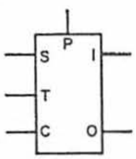


Fig. 14-9. NOR gate.

stored in a five-stage shift register. The desired character is recognized by a coincidence circuit. The state of recognition is stored in a flip-flop. When all five characters have been received, the output relay closes.

Let us see how the register stores the letter J, which is Start-M-M-S-M-S-Stop. The first logic level seen by the register is the start pulse, a space. This (L) input is inverted to a (H) and applied to SR 5 lead (S), in Fig. 14-12. After the clock pulse, the (1) output also becomes (H), which we will define as the space condition of the flip-flop. The next signal pulse (one), is a mark, which makes SR 5 lead (S) low. The next clock pulse now does two things. At this point SR 4 sees the space condition of SR 5 and duplicates its output, making SR 4 (1) low. The

*STATE AFTER TRIGGER PULSE



S	C	(1) *
H	L	H
L	H	L
L	L	REVERSES
H	H	NO CHANGE

SYMBOL

TRUTH TABLE

Fig. 14-10. J-K flip-flop and truth table.

start pulse has been passed from SR 5 to SR 4. Also, the output (1) of SR 5 is changed to high, following the input signal. At the next clock pulse, the input is a mark (pulse 2). After this clock, SR 5 and SR 4 are in a mark condition, SR 3 in a space. The shift register now contains the start and first two information pulses of the letter J. These pulses continue to enter the register from the left. Finally, the start pulse is pushed out the right end of the register, which then contains all of the five J information pulses.

Since both (H) and (L) outputs are available from each SR stage, we can select that lead of each SR that is low for a J. Only for this J (upper or lower case) will the all-low coincidence exist. These selected low outputs are now fed into a NOR gate. Recall, that the output of a NOR gate goes high only when all inputs are low. It is the NOR gate that actually recognizes the J. The high pulse output is fed into a character-1 FF, that flips and thus remembers that the J has been received. See Fig. 14-13.

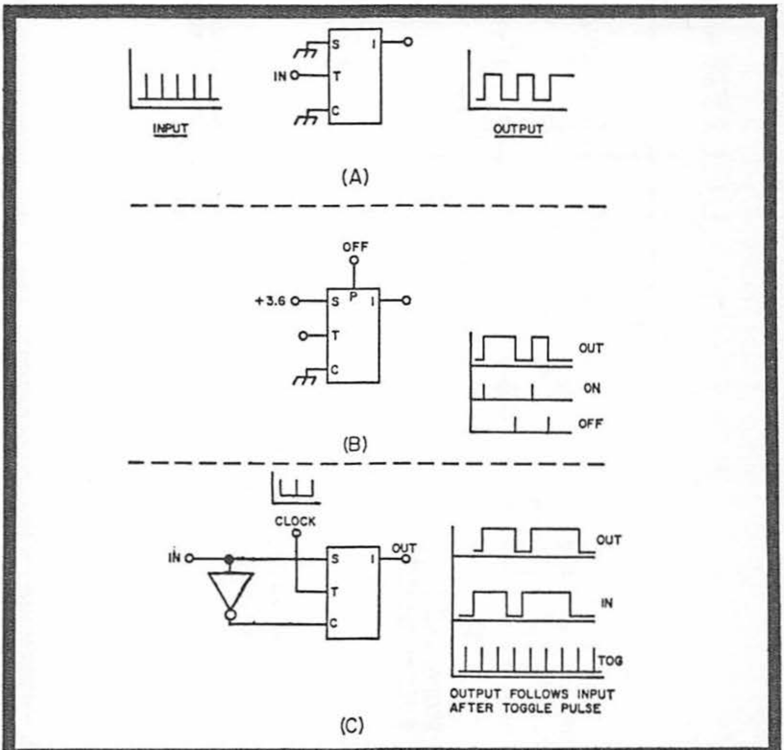


Fig. 14-11. Applications of J-K flip-flop. (A) Divider. (B) Set-Reset flip-flop. (C) Master-slave or clocked flip-flop.

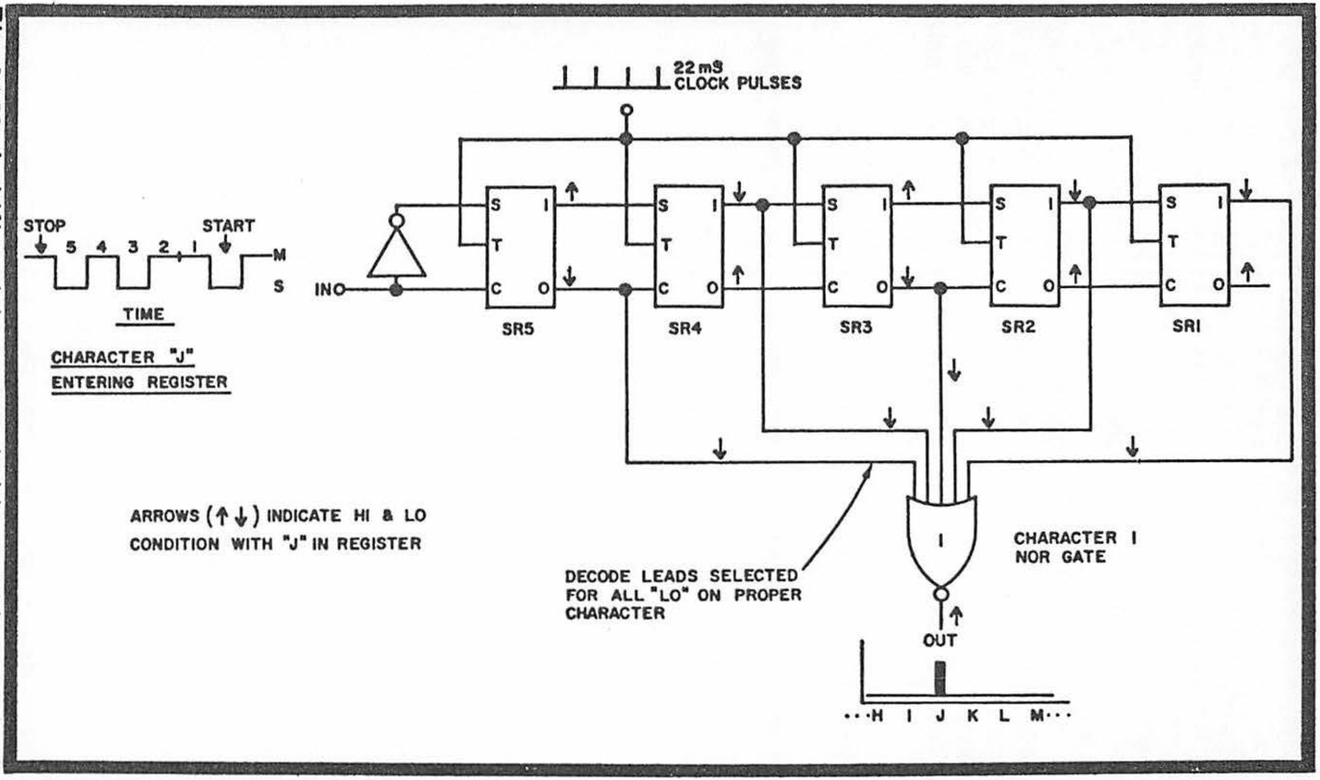


Fig. 14-12. A shift register connected to provide an output when an RTTY letter J is applied to the input.

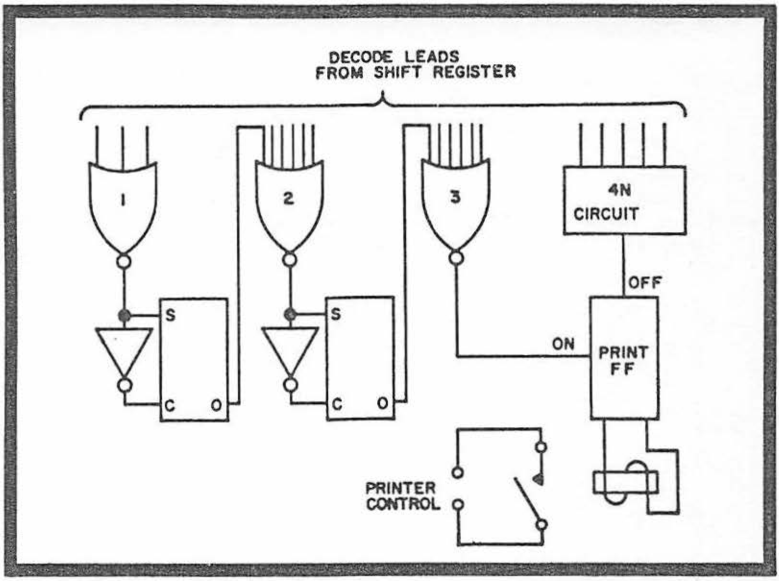


Fig. 14-13. The sequential selector. This circuitry is connected to the output of the shift register shown in Fig. 4-12 so that the letters of the call sign will turn the printer on only if they are in correct sequence.

To detect the next call letter, say K, another NOR gate is independently connected to the SR outputs that will give all lows with a K. The output of the character-1 FF feeds a low to the character-2 NOR gate so that the first character must be received before the second gate may look for its letter. This prevents the Selcal from responding to your call letters in an incorrect order.

When both the J and K have been received, the third NOR gate is free to look for the third letter, say L. When received, the third gate gives a high output which turns on the print FF and the print relay. The printer is now on and receives your message.

To turn your machine off, the sender ends the message with "NNNN." The letter N is recognized just like the J, with a properly connected NOR gate. The gate feeds a two-stage binary counter which turns off the print FF when four N's are received.

The Selcal circuit is complicated by the lack of the exact logic needed. Several NOR gates are paralleled to get enough inputs, and buffers and inverters are used to increase fan-out or driving power. Note that the A-B signal lines carry the same pulses, the split is just to prevent device overload. The ab-

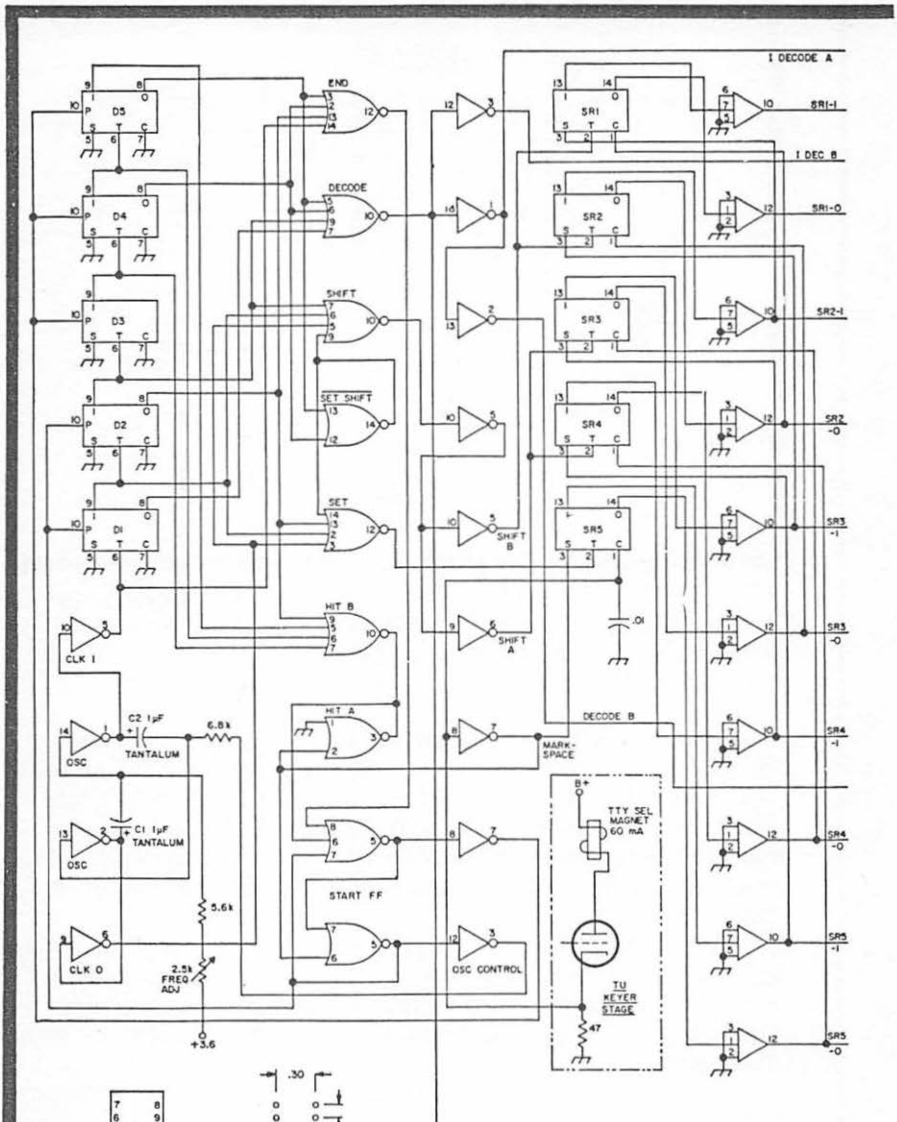
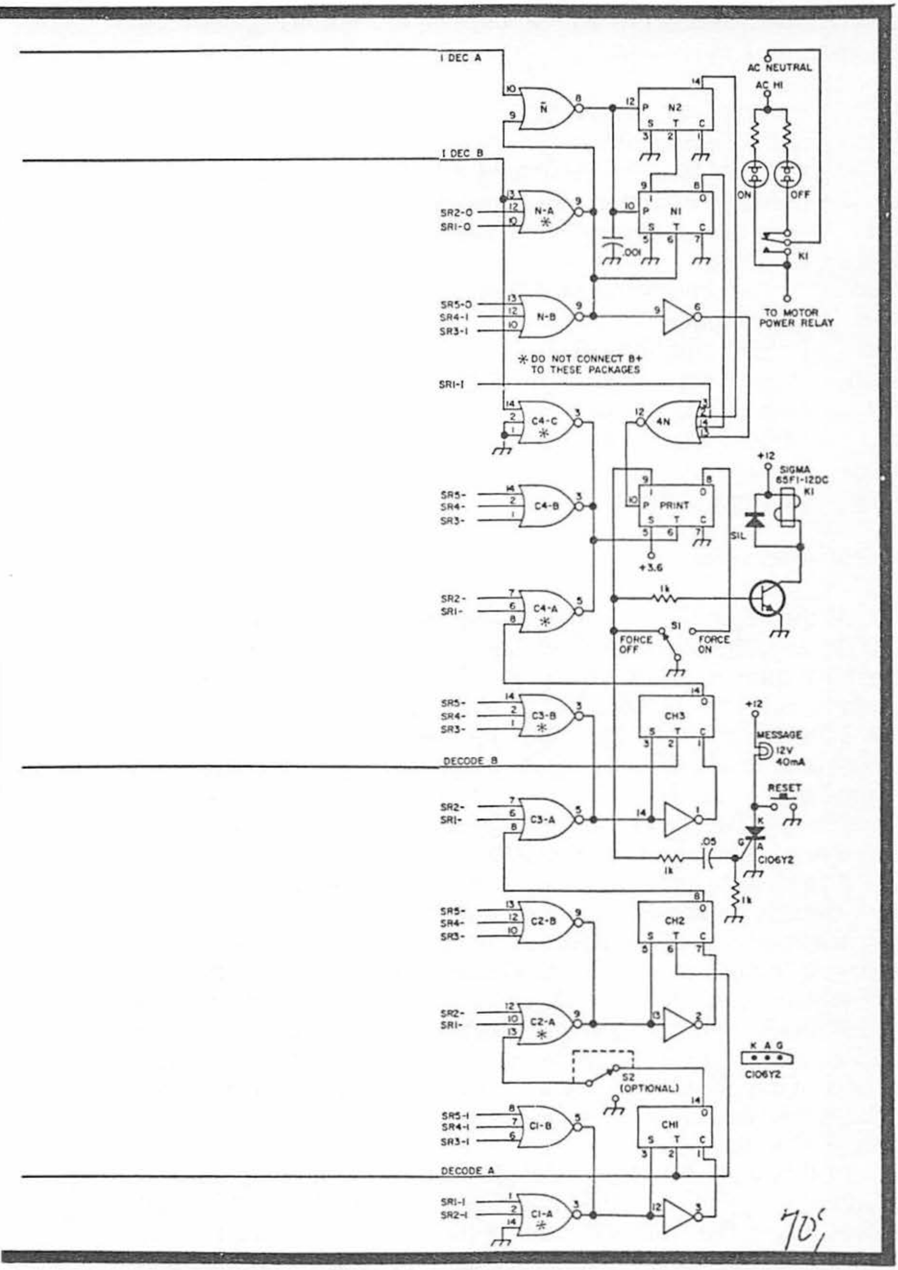


Fig. 14-14. Schematic diagram of the Selcal. Relay K1 is a 12-volt Sigma 65F1-12DC. Transistor Q1 may be any high gain silicon transistor such as Motorola MPS3393. The



message light is a Sylvania 12ES. Switch S2 is an optional "omit first character" switch.

breviations listed below will be used in the following detailed discussion of operation:

I	Inverter, or inverted.
SR	Shift register or stage.
N	Used in the 4N disconnect circuits.
C	Character; letter being recognized.
CH	Channel; memory for a character.
FF	Flip-flop.
Not	Circuit operating on all characters except ( ).
M-S	Mark-space.
Set	Pulses toggling SR 5.
Shift	Pulse toggling SR 4 through SR 1.
Hit	Non-RTTY pulses.
D	Divider stage (by two).
High	Voltage over 0.8.
Low	Voltage under 0.43.

Selcal Operation

At the beginning of a start pulse, a high occurs on the m-s line, setting the start FF (Fig. 14-14). In this set condition, the start FF places a low on the divider preset leads, allowing them to operate. The oscillator inverter raises the voltage on the 6.8K resistor to high, starting the clock oscillator. The clock is a multivibrator that generates 5.5 msec (181-Hz) square waves as shown on line 2, Fig. 14-15. These pulses are divided by two, five times, by the dividers D1-D5. The waveforms are shown on lines 3 through 7 of Fig. 14-15.

A NOR gate has a high output only if all the inputs are low. By properly selecting the clock and divider outputs, a set of low leads can be found for each single clock pulse. As an example, let us see how the single decode pulse is obtained. At the decode time, (line 11), D2 and D3 leads (1) are low, but D4 and D5 leads (1) are high. By selecting the (0) leads of D4 and D5, we obtain all low inputs for the decode gate. D3 is not needed. Only at one particular time will the above conditions exist, so the decode NOR gate gives an output pulse only at the proper decode time.

In this way, NOR gates connected to the divider outputs produce properly timed set, shift, and end pulses. The end pulses resets the start FF, ending the Selcal sequence for one character. The "reset" start FF stops for clock oscillator and presets the dividers, making them ready for the next operation. The hit gate looks for a spacing signal partway into the start pulse. If a mark exists at this time (non-RTTY

signal), the hit gate resets the start FF, terminating the operation. This resets the circuit after a false start from noise. The NOT-SET-SHIFT gate (SET SHIFT) suppresses unwanted set and shift pulses.

Now back to Fig. 14-14. Assume the call K8ERV is being received. To prevent casual copy reference to "ERV" from operating the printer, the code will be "ltrs ERV," which

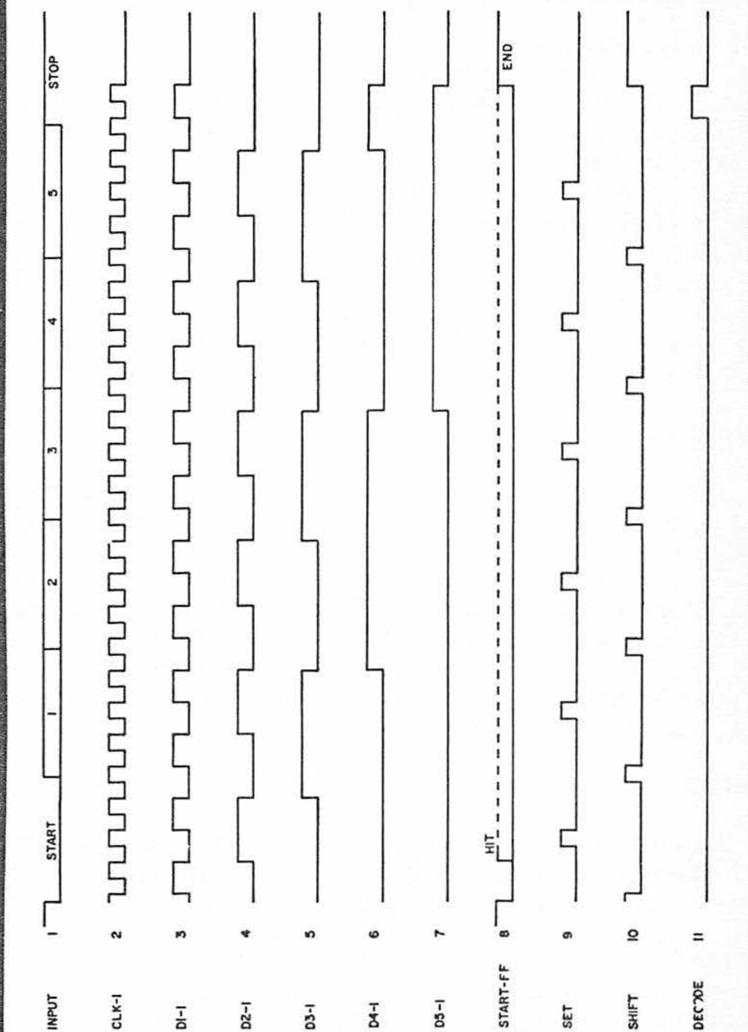


Fig. 14-15. Timing relationships for Selcal.

already exists in the callsign. The first character "letters" (ltrs) enters the shift register as described earlier. At the time of the decode pulse, the "ltrs" marks and spaces are contained in the register. The C1 NOR gates are connected to the five SR output leads that will give all lows. The C1 gates will now recognize the "ltrs" character and give a high output to the CH1 FF. This high, with the decode pulse, causes the CH1 FF to set, remembering that character one was properly received.

The CH1 FF low output (lead 0) is fed to the character-2 NOR gate, permitting it to look for, and recognize the next character, E. As the decode pulse transfers the E recognition into the CH2 FF, it also resets the CH1 FF, which insures that characters will be recognized only in the proper sequence. The E makes the CH2 FF output 0 low, and the following R makes the CH3 FF output low. This low, plus the SR lows from the V, and the inverted decode pulse (a low pulse) place all low inputs on the C4 gates. The high output from C4 sets the print FF, turning on the output relay and your printer. The Selcal has recognized the last four characters of the call K8ERV! Any wrong character will interrupt the sequence and reset the logic, preventing turn-on.

Turn Off

The print FF will now remain on until reset by a switch or by the reception of NNNN, a commercially used disconnect sequence. This section operates by recognizing and counting consecutive N's. The fourth N received gives an output through the 4N gate which resets the print FF. Any character other than N operates the NOT-N ($\overline{N}$) gate which resets the FF's, destroying the count. The Selcal must see four consecutive N characters (or upper case equivalent) somewhere in a sequence, to turn off.

All-Call

An interesting refinement will permit all Selcals to turn on with one particular calling code besides your selected call letters. Since recognition circuits exist for both "ltrs" and "N," an all-call code requiring a minimum of additional logic is "LtrsNLtrsNLtrsN." This code, besides being the easiest, will not occur in normal text. The use of six characters decreases the chance of false turn-on from noise.

Fig. 14-16 shows the all-call addition. This is a counting arrangement similar to the 4N turn-off, except that the

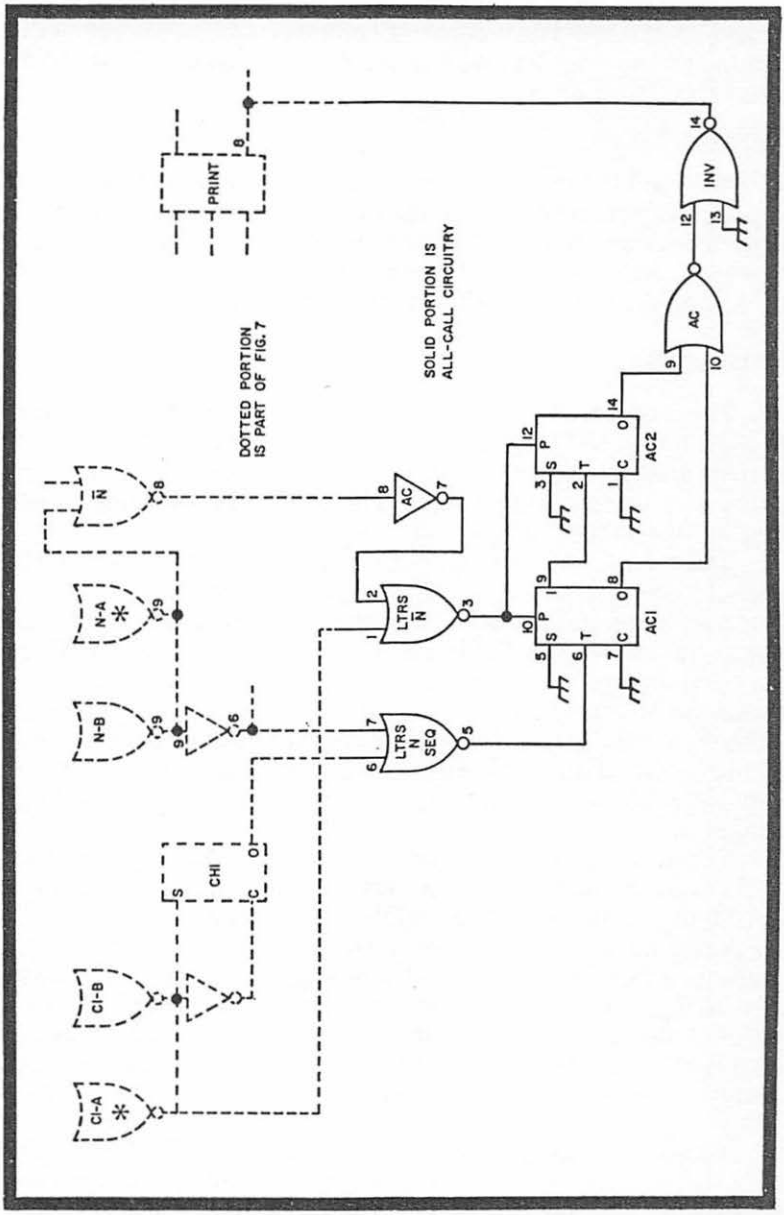


Fig. 14-16. All-call circuitry which may be added to the basic Selcal shown in Fig. 14-14. This circuitry permits the printer to be turned on by sending Ltrs N Ltrs N Ltrs N. This is particularly useful for turning on all the machines of a RTTY set.

sequence "LtrsN" is counted until all three pairs are received, turning on the print FF. The counter is reset by any character other than Ltrs or N.

Message Light

This circuit can be included to lock on a pilot light when a message is received. This alerts the operator to look at the copy. The print FF output pulse is used to trigger a small SCR that locks on a low current lamp. The lamp is manually reset by a momentary, normally open push switch.

Construction

The integrated circuits (Fig. 14-17) used in the Selcal are the Motorola RTL (Resistor-Transistor-Logic) 700 or 800 series, in a plastic dual in-line package. These differ only in price and temperature range, The 700 types covering 15-55 degrees C and the 800 types covering 0-75 degrees C.

These logic blocks may be laid out in any order. While IC sockets are available, they are expensive and unnecessary. One way to mount the IC's is to drill holes in a plastic sheet, insert the IC leads in the holes, and wire to the pins. Another way is to mount the blocks on their backs, using an adhesive, or double faced tape, and again wire to the pins. Leave plenty of room for the wires—there are several hundred of them. We strongly recommend small (No. 26) colored Teflon wire to prevent soldering iron damage in the rather cramped wiring space.

The power supply must provide 3.6 volts, plus or minus 10 percent at about 600 ma. The design shown in Fig. 14-18 has excellent regulation and negligible ripple to about 90 line volts. Its performance is better than needed but not expensive, Z1 is a group of forward-biased diodes of any silicon type, used as a low-voltage zener. Z1 is optional, being a group of one-ampere diodes used to limit the voltage in case of any type of supply failure. The Selcal can be operated from two flashlight batteries for testing, using a voltmeter in place of the output relay.

The power supply and front panel layouts (Fig. 14-19) are not critical. The only controls really needed are the on and off switches, but all sorts of pilot lamps and other accessories can be added.

Decoding

Setting up the letters you wish to receive is done by hooking the particular character NOR gates to the proper SR

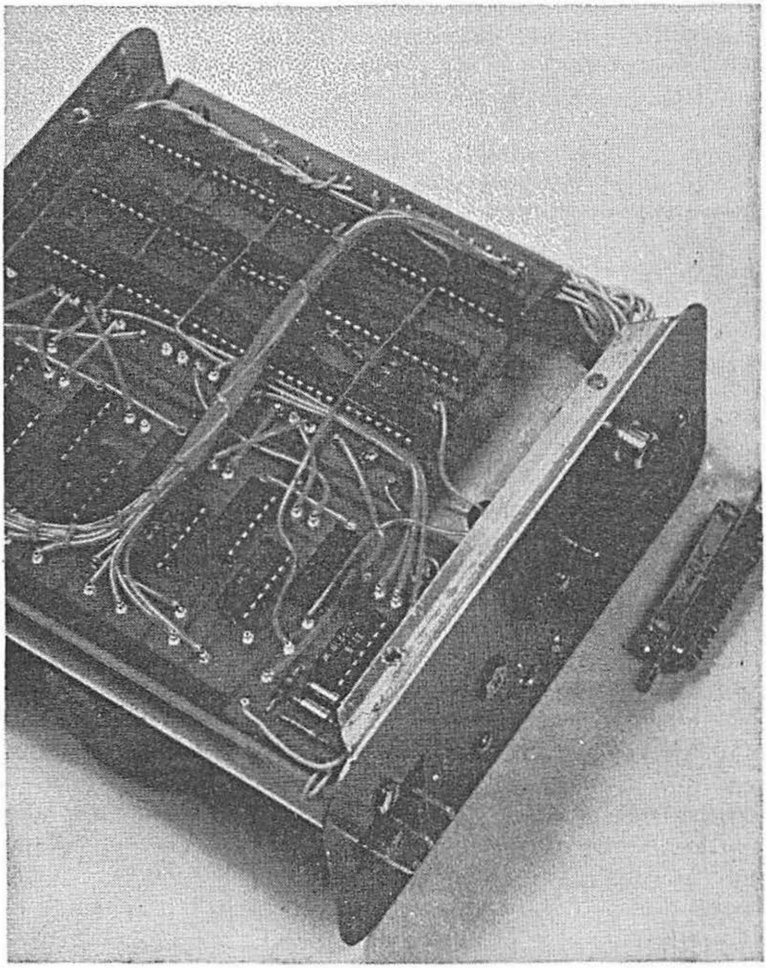


Fig. 14-17. Construction of the Selcal. The connector and plug are used for decoding purposes as described in the text.

outputs. Character 1 is shown set up for Ltrs. The N gates are, of course, wired for N's, although any repeated character could be used. To construct the decode chart (Fig. 14-20) for any character, replace the character marks with 1's and spaces with 0's, omitting the stop and start pulses. The first information pulse (after the start pulse) will eventually be in SR1, so the chart is actually reversed from the normal character construction. Since N is S-S-M-M-S, it becomes SR1-

(0), SR2-(0), SR3-(1), SR4-(1), SR5-(0). Enter your letters in rows C2, C3, C4. Now transfer this decode to the C2, C3, C4 NOR gates in Fig. 14-14. C1 is done for you for Ltrs. Connect each NOR gate lead to the indicated SR output lead. The SR outputs may feed more than one NOR input. This is the reason the non-inverting buffers are used.

The simplest method of decode wiring is to permanently connect the decode leads. But two other methods are more versatile.

Fig. 14-21 shows how twenty inexpensive slide switches can be used to set in the four characters at will. This scheme permits fairly rapid changes in the decode set. A piece of cardboard with holes that accept the slide levers in a particular decode setup can be used to check the settings.

A still faster decode change can be obtained by using a multi-pin connector as a patch board. Each decode group is wired to a separate plug and inserted into the socket in the Selcal. Twenty-five pins are required for a three-letter decode, thirty pins for four letters.

The Selcal turns on the printer motor when its code set is received. If fed with continuous random noise, eventually the Selcal will receive its code and give an unwanted turn-on. A three letter decode for commercial or experimental copy can be obtained by grounding the C2 output lead, as shown in Fig. 14-14. This is not recommended for unattended copy due to the increased possibility of noise turn-on. We suggest the Selcal be

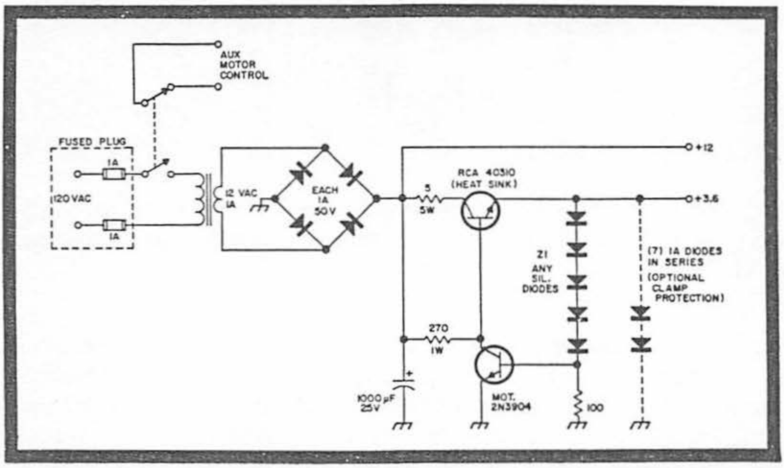


Fig. 14-18. The power supply for the Selcal. This unit provides good regulation even when the line voltage dips down to 90 volts.

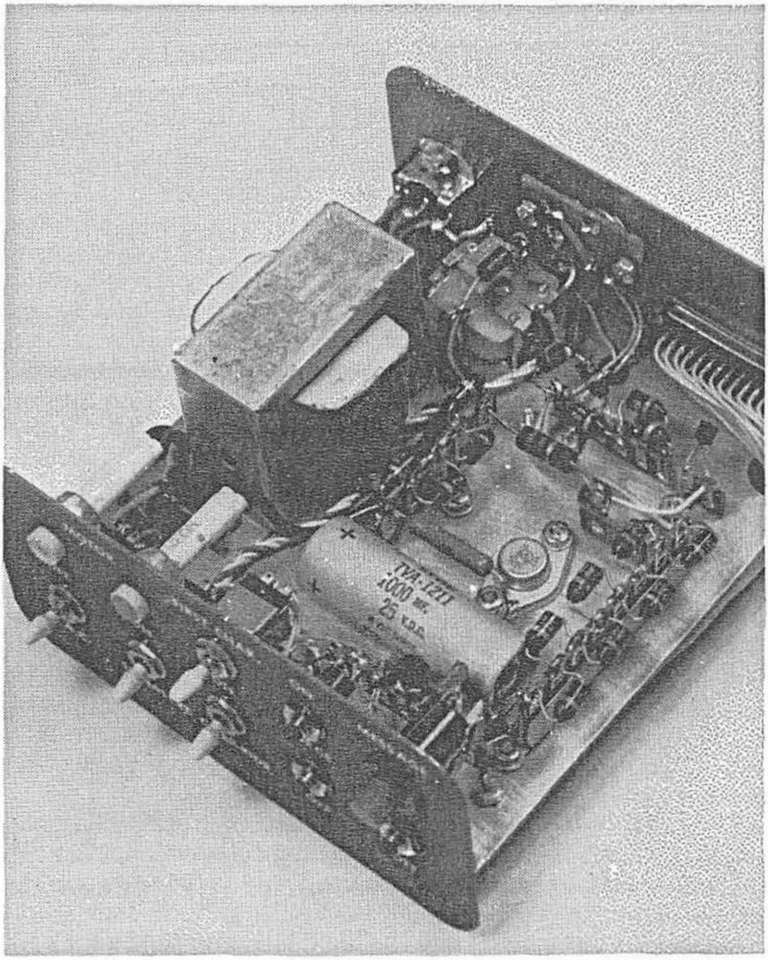


Fig. 14-19. Power supply and front panel.

teamed up with an auto-start system to inhibit the noise fed to the input.

Adjustment

The only adjustment is the clock oscillator frequency. Temporarily turn on the clock by shorting the Selcal input. Connect a scope to either clock output. Using the line frequency for comparison, adjust the 2.5K pot for 180-Hz output. If a scope is not available, set the pot in the center of the range that gives proper Selcal operation.

DECODE CHART					
SR	5	4	3	2	1
LTRS	1	1	1	1	1
C2					
C3					
C4					
N	0	1	1	0	0

Fig. 14-20. The decode chart which is used in setting up the Selcal for receiving specific call letters.

The power supply output should be from 3.3 to 3.9 volts. It can be varied slightly by varying the 100-ohm resistor between 50 and 200 ohms. If greater shift is needed, change the number of diodes in Z1. CAUTION: Do not operate into the logic with Z1 disconnected. A 6-ohm, 5-watt resistor can be used as a supply load to simulate the Selcal when tuning up. If a Variac is available, run the line voltage down until the output starts to drop. This should be about 90 volts, but depends on the gain of the 40310. Lowering the value of the 270-ohm resistor will reduce the required input voltage, but too low a value will reduce regulation and may overload the 2N3904.

If wired correctly, the Selcal should operate when connected as in Fig. 14-7. Note that the Selcal relay will not handle a printer motor load, and must be used only to drive a suitable motor relay.

Connect your printer into the local loop and send your call letters. The Selcal relay should turn on. If it does, you have made about 350 proper connections! Now send any letter except N, to reset the all-call, and then send NNNN. The Selcal relay should turn off. If it doesn't, do not despair, a troubleshooting guide follows.

Troubleshooting

First check to see if the start FF and clock oscillator are being keyed. Hook a scope or headphones through a 1000-ohm isolating resistor to the Clk-1 output. Sending any letter should produce a burst from the oscillator. Ground the Selcal input and check for proper outputs from each divider and from the set, shift and decode gates, as shown in the timing chart. Any

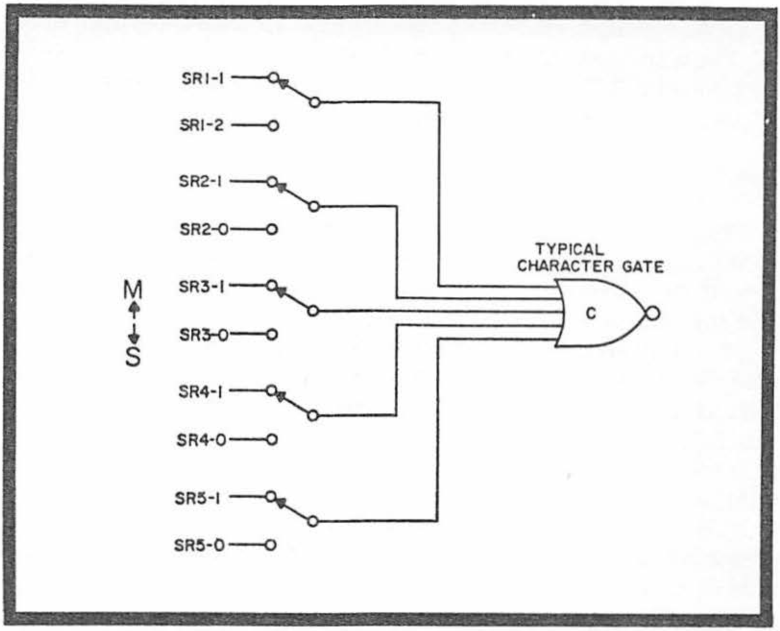


Fig. 14-21. By using slide switches in the input to the character gates, various turn-on codes may be used with the Selcal.

logic block pin, except B+, may be connected to B+, or grounded, to force a circuit on or off, without harm to the logic.

Now check the shift register by sending a "letters" character. With a voltmeter, see that all the SR-1 leads are low (less than 0.43v), and that all the SR-0 leads are high (over 0.8v). Send an N and check for lows on SR3 (4 and 1) and on SR's 1, 2, 5 (0). Any letter should leave its proper pattern in the shift register. If the higher number SR stages work, but the lower ones do not, check the wiring and logic at the point of signal loss.

Now send any letter not in your code set. A meter should show highs on all of the character FF (0) leads. Your first code character (Ltrs) should make CH1-(0) go low. The second code character will reset CH1-(0) to high and make CH2-(0) low, etc.

Check the 4N gate as follows. Send any letter other than N. All four inputs to the 4N gate should be high. The first N will place a low only on 4N pin 14. The second N should make only 4N pin 2 low. The third N will make both pins 2 and 14 low. During the fourth N, only at the decode time, do pins 3 and 13 go low, but a scope is needed to see this. The 4N gate output

briefly goes high, resetting the print FF and turning the relay off. With the exception of the 4N system, most of the Selcal functions hold their states after decode, so that a voltmeter is all that is needed for testing.

Use

In operation, the receiver, tuning unit, and Selcal are left running continuously, or connected to a time clock. The sender should transmit your call several times to insure reception and turn-on. After his call, it is helpful to include the time in GMT, followed by an extra line feed to separate the messages. After sending the message, he should return the carriage to the left, and send 8 to 10 N's. If conditions are poor, send extra N's to insure turn-off: any not needed will not be copied. Automatic CR-LF systems are very convenient for any unattended autostart or Selcal operation.

While autostart is not useful for monitoring continuous commercial stations, the Selcal is, and can be used to select only those parts of interest to you. However, for 75 or 100 WPM monitoring, the Selcal clock must be changed.

Set up the Selcal to decode the appropriate heading and you are in business. For example, the Weather Satellite predictions are preceeded by TBUS. The 4N turn-off is sent regularly.

CHAPTER

RTTY HANDBOOK

15

Accessories

This article describes several accessories for the RTTY station. The first is a regenerative repeater; you put highly distorted, biased signals in one end and get nice, clean, properly timed signals out the other. The second forms the basis of an electronic stunt box; it performs the "cleaning up" function of the first, plus converting the serial TTY signal to a 5-line parallel signal which can be used to perform various functions on receipt of a specified group of characters. The third performs all the functions of the first two plus speed conversions; with this you can put 100 wpm gears in your machine for copying the commercials and use the speed conversion function to operate at 60 wpm on the ham bands.

In order to understand the operation of these three devices, let us take a moment to review the manner in which a TTY printer decodes a signal. The start element of the code drops the selector magnet, which initiates a mechanical timing cycle. Five times during this cycle the machine mechanically samples the condition of the selector magnet and positions the code bars accordingly. In a 60 wpm machine these samples are 22 msec apart and about 4.4 msec long. The length of time between the beginning of the start element and the first sample can be varied with the range adjustment. With the range control set at 60 the samples occur 33 msec, 55 msec, 77 msec, 99 msec, and 121 msec after the beginning of the start element.

THE REGENERATIVE REPEATER

The repeater electronically samples the TTY signal in much the same manner as the printer. A start element initiates a series of seven sampling pulses. The input signal is sampled in the middle of the start element, in the middle of the five signal elements, and 11 msec into the stop element. The condition of the signal at the time of each sample is loaded into a flip-flop memory where it is stored until the next sample is taken. Thus, the output of the flip-flop is a perfectly timed TTY

signal, delayed 11 msec from the input signal, with signal elements corresponding to the condition of the input at the time of sampling (Fig. 15-1). After the stop element is sampled, the circuit resets and waits for the next start element.

The logic required to perform these functions may be implemented in many different ways. The diagrams included here show 803 series DTL integrated circuits. It should be noted, however, that the same functions could just as well be accomplished with RTL, TTL, ECL, or HTL ICs, discrete transistors, or even tubes or relays. With this series of DTL a high logic level is approximately +5v and a low logic level is approximately ground.

Referring to the logic diagram of Fig. 15-2 and the timing diagram, Fig. 15-3, a start element (space) plus NOR gate U4 input pin 1 at a low level, forcing output pin 3 and inverter U5 input pin 1 high. Output U5 pin 2 goes low, starting the 91-Hz synchronous clock.

The synchronous clock is an oscillator that can be turned on in an orderly fashion; that is, the first cycle after the start command has the same period as all the following cycles. A simple oscillator of this type is shown in Fig. 15-4. It produces a series of narrow positive-going pulses at 11 msec intervals, the first occurring 11 msec after the input goes low.

The clock output drives divider flip-flop U3, the output of which is a square wave with a period of 22 msec. NAND gate U4 picks out every other clock pulse and drives inverter U5 input pin 3. The signal at U4 pin 4 is the string of positive going sampling pulses. The first of these pulses causes J-K flip-flop U3 output pin 6, which has been high until now, to go low. As long as the K input, pin 3, of this flip-flop remains low the following sample pulses will have no effect on its output. The

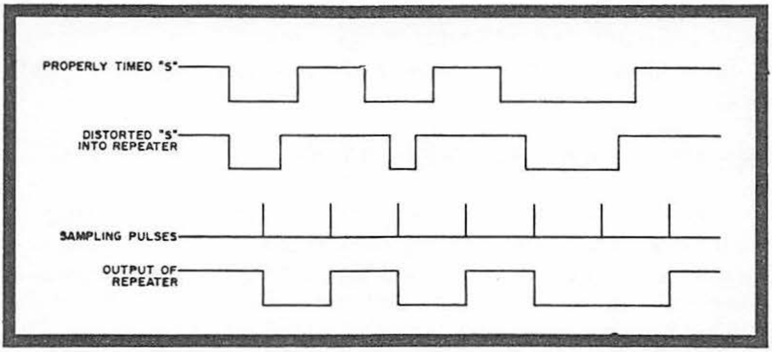


Fig. 15-1. Simplified timing diagram for a regenerative repeater.

low output at pin 6 goes to NOR gate input U4 pin 2, causing the timing cycle to continue regardless of the signal at the input. Note, however, that this happens only after 11 msec of continuous spacing signal. This means that a spacing condition must exist at the input for at least 11 msec to initiate a timing cycle. This provides protection against noise on the signal line.

The first sample pulse also loads the start element into output flip-flop U1 pin 5 and causes the counter outputs, U1 pin 9, U2 pin 5, and U2 pin 9 to step from all low (LLL), the reset condition, to (HLL). The second sample pulse loads the first signal element into the output flip-flop and steps the counter to LHL. This process continues for pulses three through six. At this time the last signal element has been loaded into the output flip-flop and the counter outputs are LHH. This makes both U4 pin 9 and U4 pin 10 high, so U4 pin 8 and U5 pin 9 go low. Inverter output U5 pin 8 J-K flip-flop U3's K input, pin 3, go high. The next sample pulse, which loads the stop element into the output flip-flop, also causes J-K flip-flop output U3 pin 6 to go high. This, together with the condition on the signal line, causes NOR output U4 pin 3 to go low, stopping the clock, resetting the counter to LLL, and forcing the output flip-flop to remain in the mark condition until the next start element initiates a new timing cycle. Just add some simple level converters to make your TU, printer, keyboard, and keyer DTL compatible and you are all set to clean up distorted received signals and to transmit perfectly clean signals from your keyboard.

THE STUNT BOX

The stunt box is a relatively simple expansion of the repeater, requiring only a few more parts and the rerouting of a few wires. Comparing the stunt box (Fig. 15-5) and the repeater, we find three differences:

First, the output flip-flop is now the first stage of a six-stage shift register; when the first signal element is loaded into U1 pin 5, the start element moves to U1 pin 9, and so on until after the sixth sample, when the start element is present at U6 pin 9, and five signal elements are stored in the first five stages of the register.

Second, instead of counting out seven sample pulses with a counter, we now wait for the start element to move into the last stage of the register, causing U6 pin 8 and U3 pin 8 to go high. Now, as before, the seventh pulse samples the stop element and resets the entire sequence.

Third, 11 msec after the sixth pulse we find the U5 pin 9, U5 pin 11, and U5 pin 13 are all low, allowing U5 pin 8, 10, and 12,

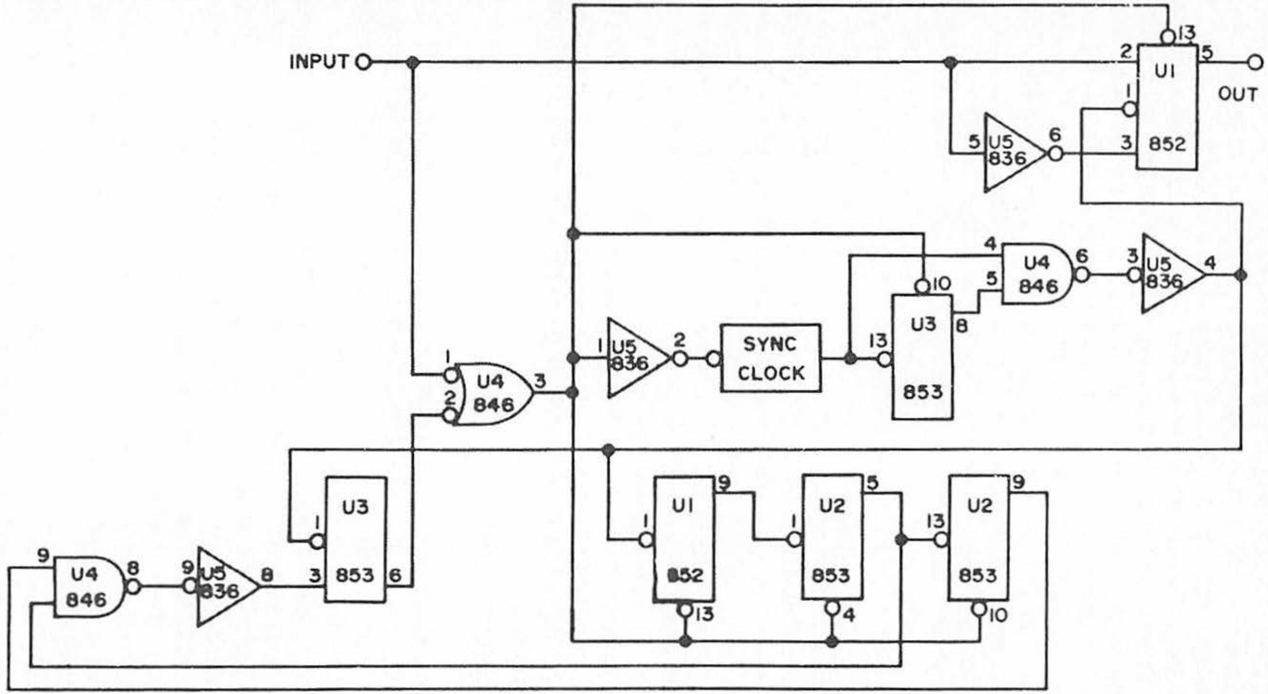


Fig. 15-2. Logic diagram for regenerative repeater.

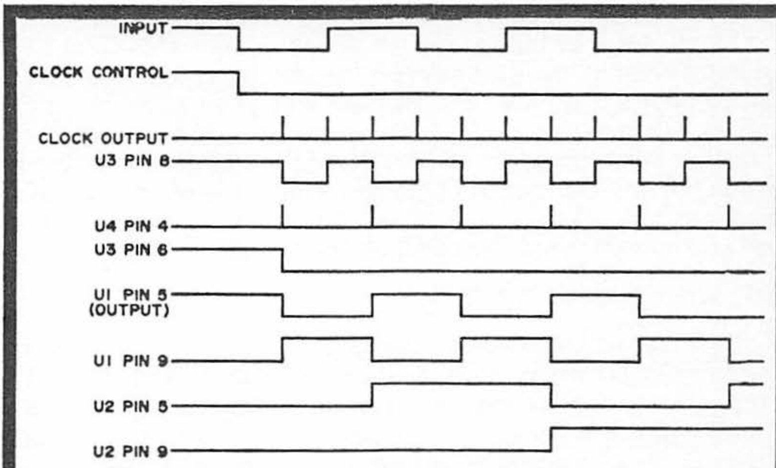


Fig. 15-3. Timing diagram for circuit shown in Fig. 15-2.

and U4 pin 12 to go high and forcing U4 pin 11 low. Thus, we get both a positive and a negative strobe pulse which occur once each character, and at a time when the entire character code is stored in the register. This is all the information we need to decode a character.

A simple stunt box decoder is shown in Fig. 15-6. This decoder is set up to respond to the two character sequence ZB. When the positive strobe pulse occurs, the two flip-flops sample the NAND gate outputs U7 pin 6 and U7 pin 8. If the character in the register is anything other than a Z, U7 pin 6 will be high when strobed; and U8 pin 6 will remain low. A Z forces U7 pin 6 low, and U8 pin 6 goes high after the strobe

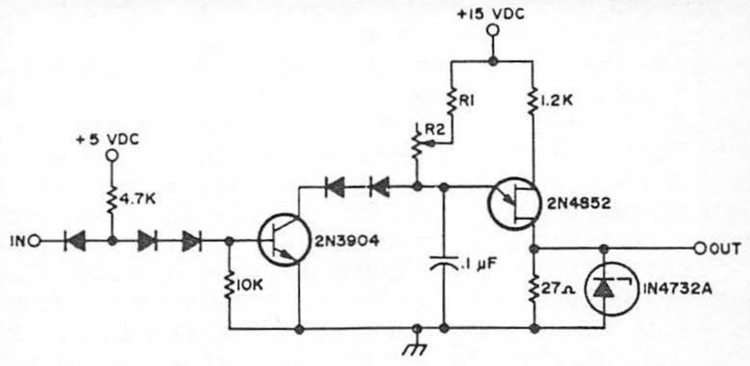


Fig. 15-4. Synchronous clock.

pulse. If the next character is anything other than a B, U7 pin 8 and U7 pin 6 will be high at the time of the strobe pulse and U8 pin 8 will remain low. If, however, the character after the Z is a B, U7 pin 8 will go low; and U8 pin 8 will go high and remain high until the next character is received. Using this decoder as a starting point, much more complex decoders can be built to respond to any number of character sequences of any length. These can be used to set and reset latches to turn your printer, tape unit, or coffee pot on and off on command.

THE SPEED CONVERTER

Now that we have taken a 60 wpm TTY signal and shifted it into a register where we have it temporarily stored, all that remains to be done to make it print on a 100 wpm machine is to load it into another resistor, in parallel form, and shift it out to the printer with shift pulses that occur every 13.5 msec. The hardware required to accomplish this is shown in Fig. 15-7.

Before a character is received U6 pin 8 and the input to the 74.2-Hz synchronous clock are low, the clock is running, and the register is shifting. The input to the first stage of the register, U10 pin 3, is held low and the output to the printer, U14 pin 6, remains high, or marking. When the start element of a character shifts into the last stage of the first register, U6 pin 8 goes high and the 74.2-Hz clock stops. When the strobe pulse occurs, 11 msec later, U5 pin 10 goes high, loading the five signal elements into the first five stages of the second register. At the same time U4 pin 11 goes low, forcing a space, or start element, into the last stage of the second register. The output to the printer remains in the marking condition, however, since U6 pin 9 is holding U14 pin 4 and the input to the output NAND gate at a low level. When the stop element has been sampled and the first register resets, U6 pin 9 goes high, allowing the start element in the last stage of the second register to appear at the output to the printer. At the same time U6 pin 8 goes low, restarting the 74.2-Hz clock and allowing the character to shift out of the register to the printer. As the character shifts out, the register fills up with marks so that 13.5 msec after the last signal element is sent to the printer the output goes to a marking condition and stays there until the next character arrives.

To convert the 100 wpm output of the keyboard to a 60 wpm input for your keyer, it is only necessary to reverse the input and output leads of the converter and switch the clock frequencies from 91-Hz and 74.2-Hz to 148.4-Hz and 45.5-Hz. Of course, the keyboard may only be operated at typing speeds up to 60 wpm.

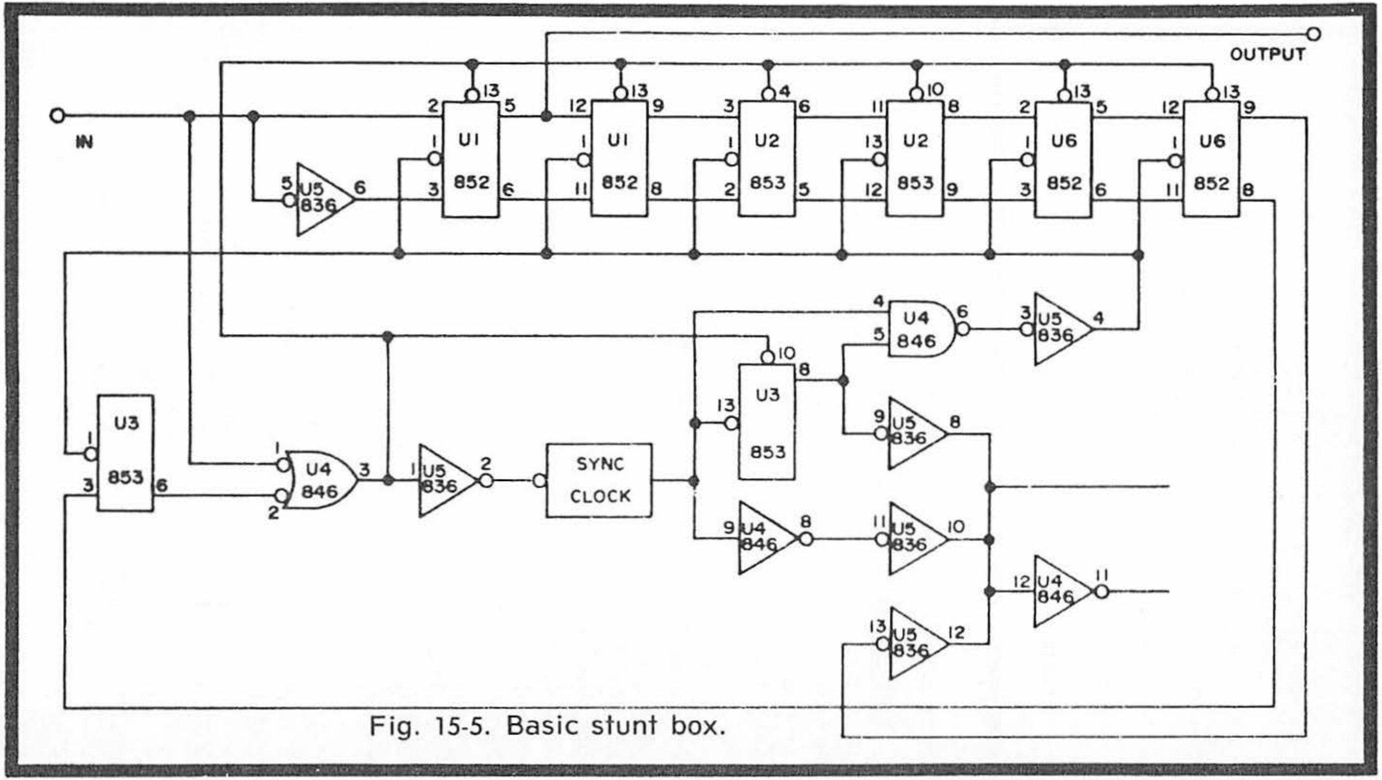


Fig. 15-5. Basic stunt box.

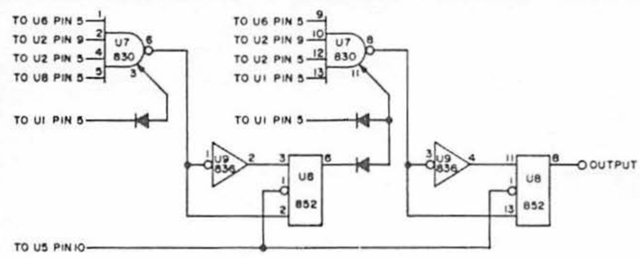


Fig. 15-6. Simple stunt box decoder.

TT-63A AS A DISPLAY GENERATOR

Teletype Repeater TT-63A-FGC (Fig. 15-8) is an inexpensive and widely available piece of surplus equipment. Although many amateurs have purchased this unit, problems in getting it into operation have resulted in most TT-63A's gathering dust on a shelf or filling up useless space in relay racks.

A novel use for the TT-63A is as a Teletype character-synchronized sweep generator. The TT-63A recognizes a Teletype character as part of its basic function. This characteristic is used to generate a scope presentation in which the Teletype character is always perfectly synchronized, and time marks provided, at either keyboard or machine speeds. The basic concept is applicable to any other surplus Teletype equipment having this recognition capability.

Modification should cost less than $5, depending on the state of your junk box, and should not take more than several evenings to build.

A few comments are in order. Attempting to convert this unit to a TU by adding a standard limiter-discriminator front end will not be particularly successful. This is due to two design peculiarities: The TT-63A is extremely susceptible to errors resulting from pulse splitting and the 1 msec sampling pulses. It needs a very good low-pass filter between the discriminator output and the trigger input. By the time you have added a truly effective low-pass filter, you will have probably duplicated half a TTL without its benefits. Also, the chassis is too crowded for the required extra components. Another unfortunate characteristic is its performance at machine speeds when a start pulse is lost: the TT-63A prints

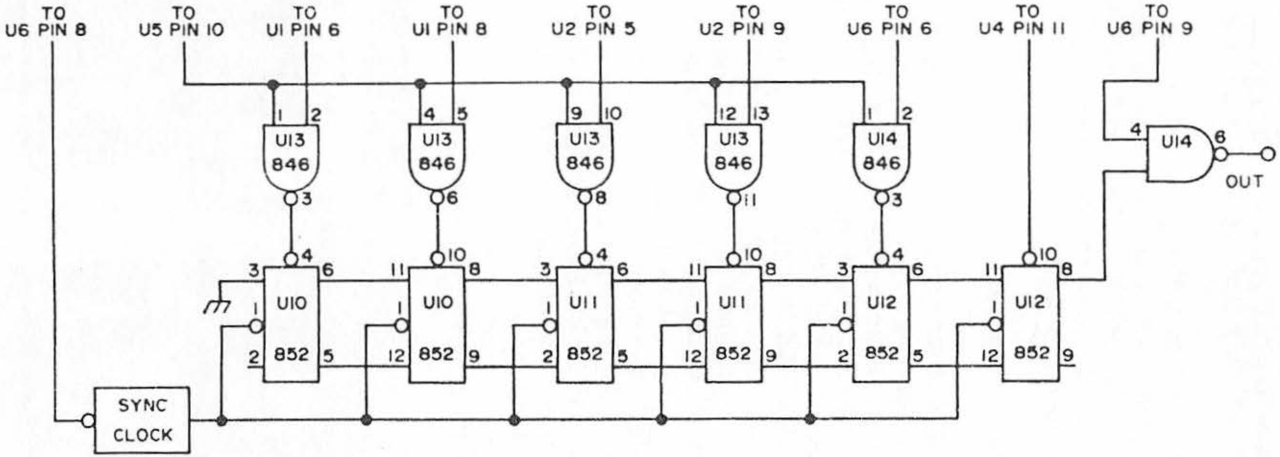


Fig. 15-7. Speed converter adapter for stunt box.

several erroneous characters before timing synchronization can be regained. In addition, if you are not using coax to your receiver, you may possibly pick up hash from the keying relay.

None of these problems will significantly affect its performance as a synchronized character display generator. Adding the simple circuits shown in Figs. 15-9 and 15-10 and slightly modifying your TU is all that is required to make the TT-63A a truly useful RTTY accessory.

The Bootstrap Circuit Operation

Normally, the grid of the 6SN7 is biased positive, so that the tube is fully conducting. This effectively removes the charge across C102. When the negative 50v gate from V6 cuts this tube off, C102 begins charging through D101 and R102. The cathode follower portion of the 6SN7 feeds back the output ramp to the high end of R102 to maintain a constant charging rate and provide a linear sawtooth. With the components shown, the output is approximately 45v. The value of C103 is not too critical since the charging of C102 occurs mostly in its linear region even without bootstrapping. Removing C103 affects the amplitude far more than the linearity. It is necessary, however, to use a good quality capacitor for C102—**do not use an electrolytic!** R103 allows the output signal to be balanced around ground. Some slight loss of signal results from this arrangement, but it is effective and allows the scope to be switched from a cross tuning pattern to this pulse analysis mode without making centering adjustments on the

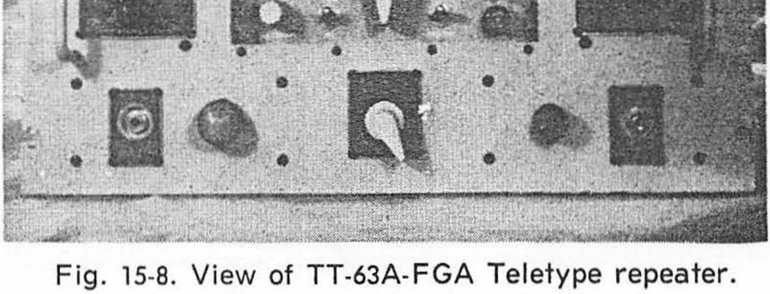


Fig. 15-8. View of TT-63A-FGA Teletype repeater.

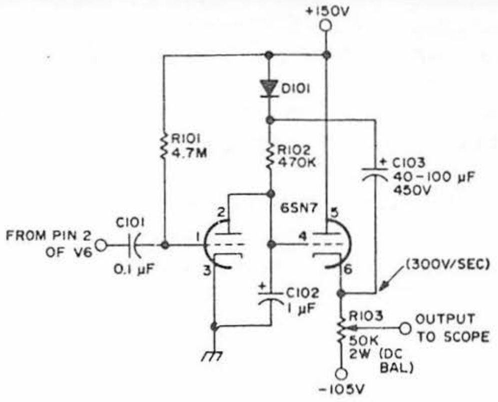


Fig. 15-9. Bootstrap sweep generator.

scope. It should be noted that the start pulse is drastically shortened on the actual display. This is due to the poor risetime of the leading edge of the gating pulse.

Construction

The bootstrap circuitry is placed where V1, V2, and V3 were originally located. These tubes were part of the original tone portion of the repeater and are not used. The original 6SN7 (V1) is used as the bootstrap tube. One of the 6H6s may be used in place of the semiconductor diode shown in the schematic. The remaining sockets are handy as tie points for the bootstrap components. Be careful when trimming wires on these sockets, as several terminals are used for tie points for other stages. The sweep centering pot is a screwdriver ad-

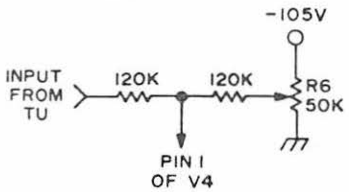


Fig. 15-10. TT-63A input modification to provide negative mark voltage for proper operation of V4. R6 is the input attenuator which is already in the unit.

justment type. It can be located on the left side of the chassis after removing some unused tone parts.

The pickoff for the time-marked pulse display is shifted from its original point to the junction of resistors R55, R56, and R57. This change increases the amplitude of the time markers over the original point. A slight negative DC bias is introduced on the display by making this change, but the resulting improvement in marker visibility is worth this slight inconvenience. The resulting display has vigorous negative pips where the pulse transitions should occur and lower amplitude positive sampling pulses. The range control centers the received pulses between these negative markers.

Connecting the TU

For proper operation of the repeater, the input to the Schmitt trigger must go negative on the stop mark. A few additional parts can take care of this requirement. Since most TU's can be readily modified to provide positive mark outputs; a simple inverter, shown in Fig. 15-11, is added to provide positive space pulses. The new repeater input circuit shifts the resulting zero-volt mark output from the inverter to a negative mark (Fig. 15-12). Measurements show that the Schmitt trigger turns on at -2.75v and off at -2v. To get the unit in operation, rotate R6 to the position which gives the best blinking of the neon. This setting is not critical.

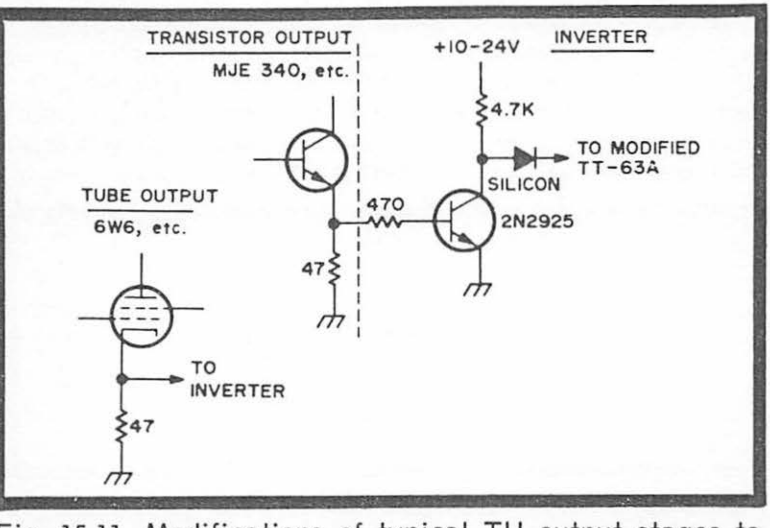


Fig. 15-11. Modifications of typical TU output stages to provide proper polarity pulses to the TT-63A input.

Usage

When this system is set up (Fig. 15-13), you can readily observe distortions on incoming signals, dirty contacts, the poor efficiency of your lowpass filters, etc. If your keyboard is in series with your selector magnets, you can readily check your own contacts. When this was first tried on my transistor TU, which had only 24v available to drive the selector magnet, the observed pulse distortion was appalling. Going to a high-voltage loop made the pulses look as they should. The vertical input to the scope can be connected to various areas of your TU and the pulses chased through always in perfect sync. This is a truly enlightening experience.

Use a 5-inch scope for monitoring. Anything less will make observing the pulses too difficult. A DC scope must be used in order to pass these relatively slow waveforms without distortion. The pulses are best viewed if the scope gain is not set too high. They are easiest to watch when they are almost square on the screen.

A very nice feature of this system is that you can tell if you have a legitimate RTTY signal tuned in and whether it is right side up, the right speed, etc., without turning on your printer. You merely see if a synchronized sweep is being generated by the TT-63A. This is possible only with a legitimate signal. This certainly saves a lot of useless clatter, bells ringing, waste paper, etc., when tuning across the bands.

SOPHISTICATED RTTY TONE GENERATOR

This Tone Generator (Fig. 15-14) is a unique piece of RTTY test equipment. It is capable of simulating a variety of

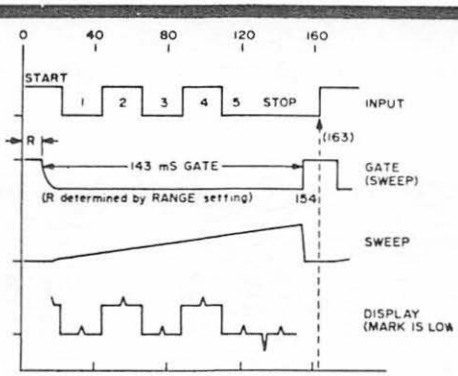


Fig. 15-12. Waveforms and time relationships.

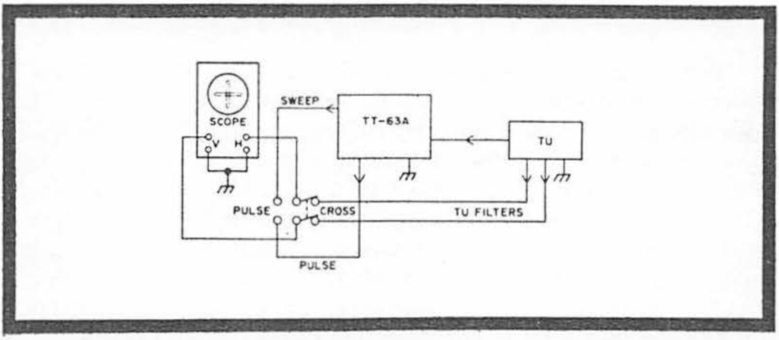


Fig. 15-13. Block diagram of typical component interconnections. The TT-63A subpanel can be rewired to do the required switching. Attenuators can be added to balance the voltages across the filters to that of the sweep and pulse voltages so that the scope gain does not need to be adjusted when switching displays.

Teletype operating and QRM modes, since it can deliver all possible combinations of the 2125-Hz mark and 2975-Hz space audio tones. When not serving in the test equipment capacity, it functions as a conventional AFSK oscillator.

It may be keyed externally by a contact closure such as from a machine keyboard, or by a 3.5-volt peak-to-peak square wave signal. An internal keying generator provides a zero-bias 22-Hz keying signal. The internal keying signal allows the unit to be used for such things as adjusting TU polar relays and is also a great convenience when chasing signals through the local system with an oscilloscope.

The unit is designed around transistors and low-cost integrated circuits. It requires 12 volts DC (plus or minus 10 percent) at about 450 ma and delivers a maximum audio output in excess of minus 10 dbm into 600 ohms. Front-panel

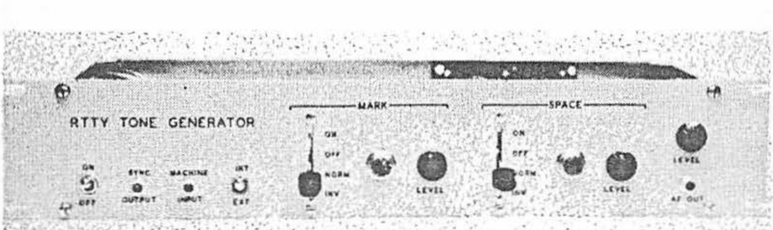


Fig. 15-14. RTTY tone generator.

controls adjust the mark-space amplitude ratio and the output level, and select the desired operating mode.

System Operation

Fig. 15-15 is a block diagram of the RTTY Tone Generator. The mark and space oscillator outputs are fed through separate level controls to independent gated amplifiers. The gated amplifiers stop or amplify their respective signal inputs depending on the information they receive from the keying logic and keying selector circuits. The audio signals from the two gated amplifiers are combined at the output level adjustment potentiometer. The signal from the potentiometer wiper is connected to the mark-space output jack, J3. The particular combination of the mark and space audio tones present at the output jack depends on the settings of S3 and S4. The following possibilities exist:

1. Mark signal off, space signal off.
2. Mark signal on, space signal on.
3. Mark signal on, space signal off.
4. Mark signal off, space signal on.
5. Mark signal keyed on and off, space signal off.
6. Mark signal keyed on and off, space signal on.
7. Mark signal off, space signal keyed on and off.
8. Mark signal on, space signal keyed on and off.
9. Mark signal keyed on and off, space signal keyed on and off—mark signal present when space signal is absent and vice versa.
10. Mark signal keyed on and off, space signal keyed on and off—mark and space signal present and absent simultaneously.

The lamps DS1 and DS2 indicate the state of the two gated amplifiers. They are extinguished when the amplifiers are in the stop mode, and illuminated when the amplifiers are in the amplify mode. The keying selector switch (S2) connects the keying logic input to either the internal keying generator or the external keying input jack. The power distribution circuits provide minus 12 volts DC, minus 3.6 volts DC regulated, and minus 8.2 volts DC regulated to the various circuits.

Circuit Description

The majority of the circuitry is assembled on three 3" x 4" homemade component boards. The mark and space oscillators (Fig. 15-16) are contained on one of the boards (CB1), the gated amplifiers and lamp drivers (Fig. 15-17) on

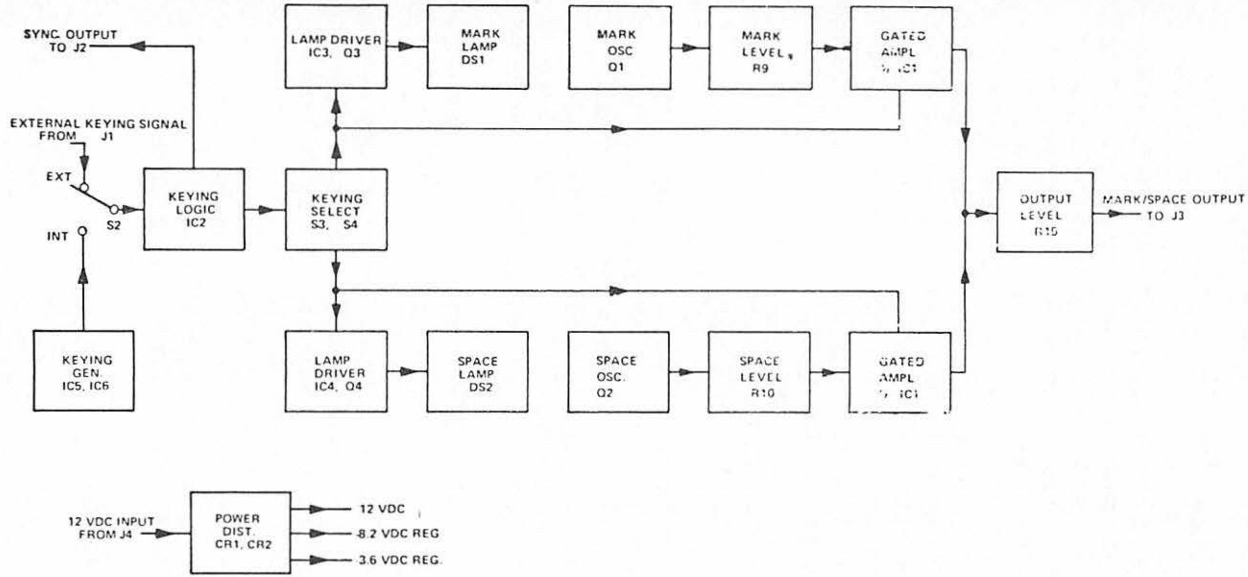


Fig. 15-15. Block diagram of RTTY tone generator.

another (CB2), and the power distribution and keying circuits (Fig. 15-18) on the third (CB3). The complete RTTY Tone Generator is shown schematically in Fig. 15-19.

The mark and space oscillators are conventional LC oscillators designed so that their operating frequencies are essentially independent of transistor case temperature. The two inductors (L1 and L2) are the usual 88-mh loading coils. Capacitors C1 through C6 are mylar dielectric units having a plus or minus 10 percent capacitance tolerance. Do not use ceramic capacitors in any of the frequency determining networks: ceramics are both temperature and voltage sensitive (Fig. 15-20). Turns are removed from L1 and L2 to adjust the oscillator frequencies to 2125 Hz and 2975 Hz respectively. This will be discussed later in the final adjustments. The capacitor values specified should resonate with any of the available 88-mh inducators.

The mark and space gated amplifier and lamp driver channels are identical. The mark signal from R9 is fed to one input of a two-input NAND gate. IC1 is a dual two-input NAND gate. Potentiometer R11 biases this input (pin 5 of IC1) so that the gate can operate as a linear amplifier. The amplified mark signal appears at pin 6 of the gate as long as pin 3 is at or near minus 3.6 volts DC. When pin 3 of the gate is at or near ground, pin 6 of the gate is essentially connected to pin 4 of the gate and the mark signal output at pin 6 disappears. The two-input logic gate thus provides a convenient means for switching (or gating) and amplifying the mark signal. The space channel functions in exactly the same manner, utilizing the two-input gate associated with plug 1, 2, and 7 of IC1.

When terminal 6 of CB2 is at or near ground potential, the mark signal output from IC1 is absent. Pin 3 of IC3 will also be at or near ground potential and the output of IC3 (pin 5) will be at or near minus 3.6 volts DC. This voltage drives Q3 into conduction, extinguishing DS1. The whole procedure is reversed when terminal 6 of CB2 is at or near minus 3.6 volts DC. The mark signal from IC1 may be present (depending on the setting of R9), the output of IC3 will be at or near ground potential, Q3 will not conduct, and DS1 will illuminate. The space lamp driver channel (IC4, Q4, and DS2) functions in exactly the same manner, responding to keying signals at terminal 5 of CB2. IC3 and IC4 are buffer amplifiers. Their sole function is to isolate the lamp driver circuits from the keying signals and assure complete switching of Q3 and Q4.

IC2 conditions the keying signal applied to terminal 14 of CB3. When the keying signal is at or near ground potential, terminal 17 of CB3 is at or near minus 3.6 volts DC and ter-

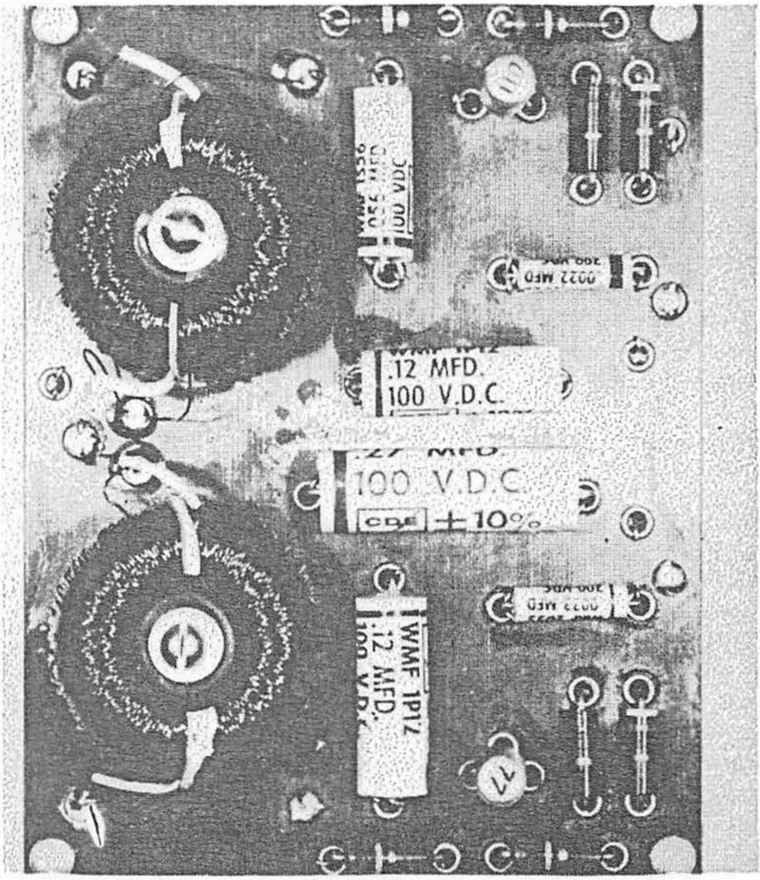


Fig. 15-16. Oscillator component board. 2125 kHz mark section is on left.

minal 16 is at or near ground potential. Conversely, when the keying input at terminal 14 is at or near minus 3.6 volts DC (or open circuited), terminal 17 is at or near ground potential and terminal 16 is at or near minus 3.6 volts DC. IC2 is a dual two-input gate operating as two inverters. Only one inverter is required to form the complementary outputs at terminals 16 and 17, but two are used to provide complete standardization of the keying signal.

One of the keying outputs is connected to terminal 15 and brought out to the front panel for use as an oscilloscope synchronizing signal. The keying mode switches (S3 and S4) select either of the two keying outputs, minus 3.6 volts DC or ground, and route the selected levels to the keying inputs of

CB2. Fig. 15-21 shows the presence or absence of the mark and space outputs with all possible keying S3 and S4 combinations.

A third dual two-input gate (IC5) is connected as an astable multivibrator that forms the time-base for the internal keying signal. One of the primary requirements of the internal keying signal is that both halves of the cycle be of exactly the same time duration. If both sections of the gate were identical, if C11 and C12 were identical, if R18 and R19 were identical, and if the multivibrator were not connected to an external load, its output would be time-symmetrical and therefore suitable as the zero-bias internal keying signal. None of these

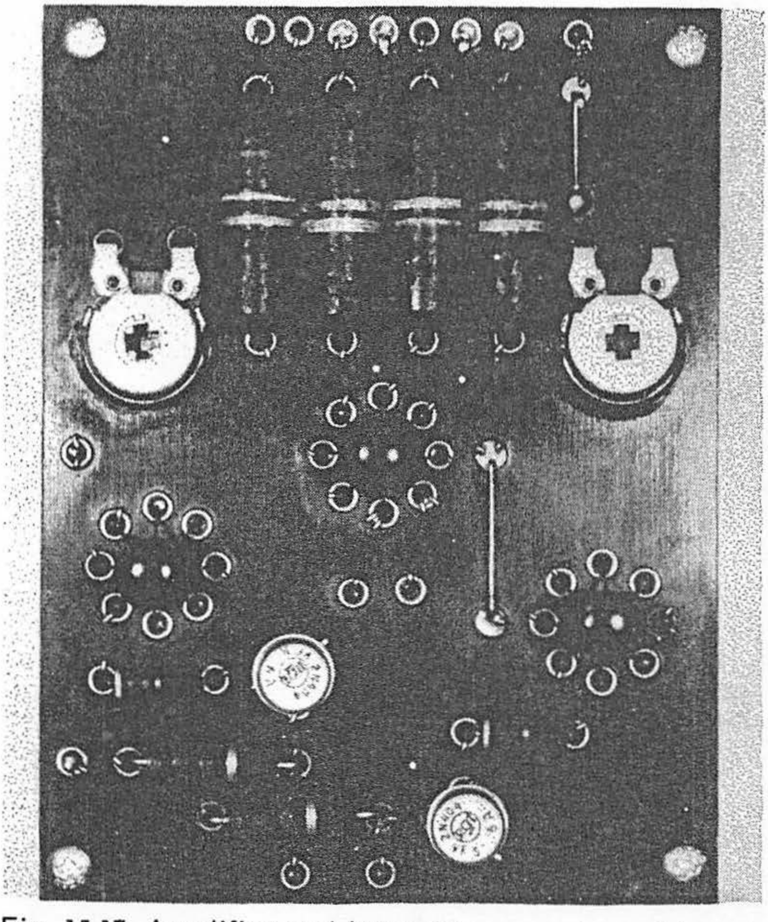


Fig. 15-17. Amplifier and lamp-driver component board. The four large capacitors could be replaced with disc ceramics.

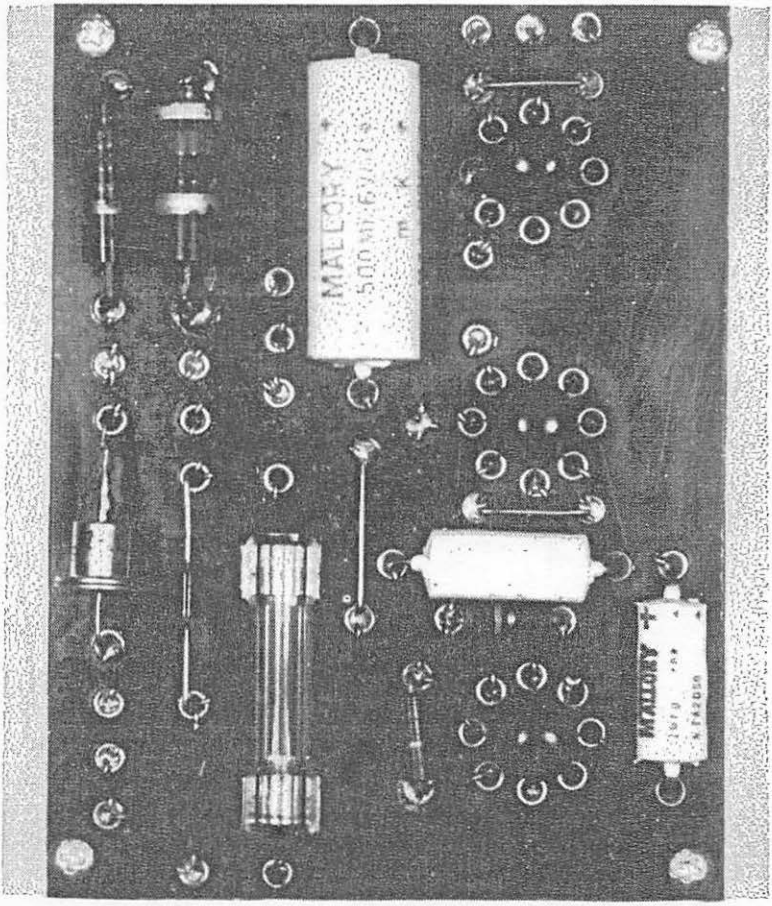


Fig. 15-18. Power distribution board.

"IF" conditions are readily met in practice. There is, however, a simple solution to the problem.

IC6 is a J-K flip-flop. Connected as shown, its output (pin 7 in this case) changes state each time the input (pin 2) switches from 0 to minus 3.6 volts DC. When the input switches back to 0 from minus 3.6 volts DC, the flip-flop does not change state. Bear in mind that 0 and minus 3.6 volts DC are only nominal values and that the flip-flop sense only HI (positive) to LO (less positive) transitions of the input signal. In each complete cycle of the multivibrator output, there is only one HI to LO transition. When the multivibrator output is connected to the flip-flop input, the flip-flop output changes state once for every complete cycle of the multivibrator output. The time duration

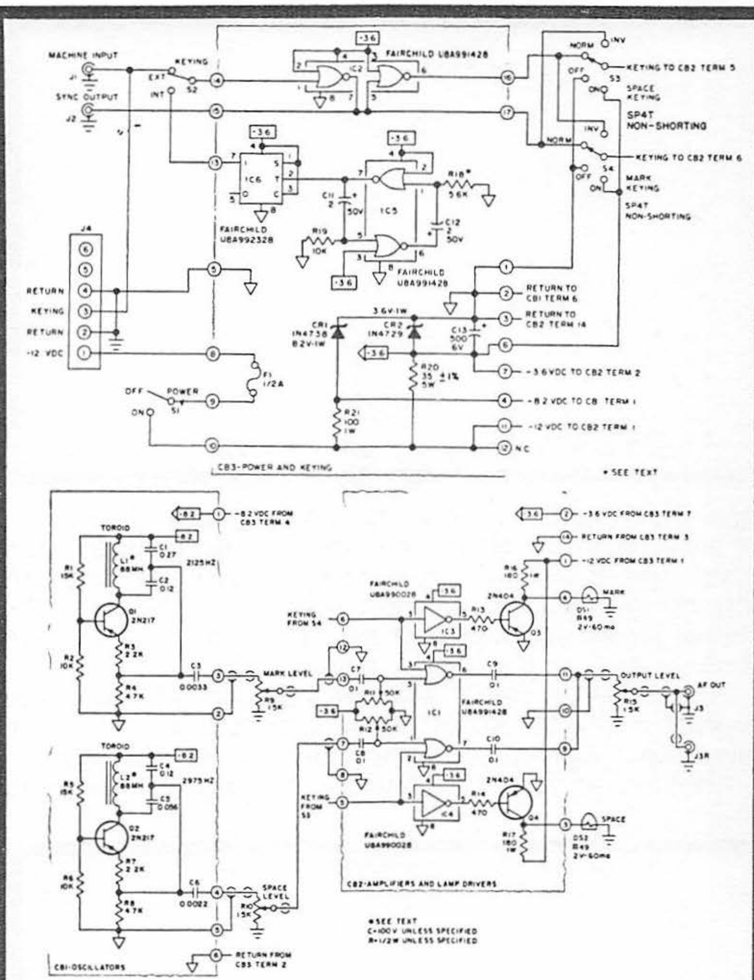


Fig. 15-19. Schematic of RTTY tone generator.

of each half cycle of the flip-flop output is equal to the time duration of one complete cycle of the multivibrator output. If the flip-flop output is used as the internal keying signal, then the internal keying signal is time-symmetrical regardless of how unsymmetrical the multivibrator output is. This is illustrated graphically in Fig. 15-22. The multivibrator must operate at 44 Hz to provide 22 Hz at the flip-flop output.

The internal 22-Hz zero-bias keying signal from the output of IC6 is connected to switch S2. S2 selects either the internal or external keying signal and applies it to the keying logic

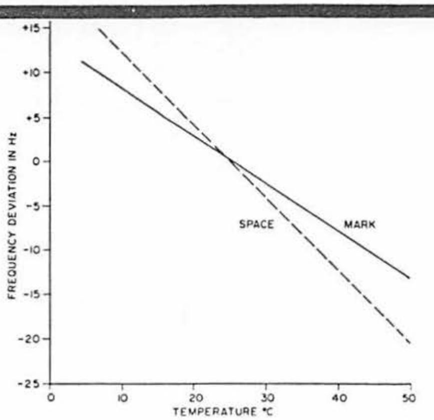


Fig. 15-20. Temperature effects upon frequency.

input. Referring to Fig. 15-21, the internal keying signal has the same effect as alternately opening and grounding the keying input. When the internal keying signal is at or near zero volts, the "keying input grounded" columns apply. When the internal keying signal is at or near minus 3.6 volts DC, the "keying input open" columns apply.

The minus 8.2 volts DC and minus 3.6 volts DC sources are derived from the 12 volts DC plus or minus 10 percent input by conventional shunt zener diode regulators. Use the resistor and zener diode values and tolerances specified in the parts list.

selector switch positions		keying input grounded		keying input open	
mark	space	mark	space	mark	space
on	on	1	1	1	1
on	off	1	0	1	0
on	norm	1	0	1	1
on	inv	1	1	1	0
off	on	0	1	0	1
off	off	0	0	0	0
off	norm	0	0	0	1
off	inv	0	1	0	0
norm	on	1	1	0	1
norm	off	1	0	0	0
norm	norm	1	0	0	1
norm	inv	1	1	0	0
inv	on	0	1	1	1
inv	off	0	0	1	0
inv	norm	0	0	1	1
inv	inv	0	1	1	0

'0' indicates signal absent
'1' indicates signal present

Fig. 15-21. Mark and space outputs as functions of switches S3 and S4.

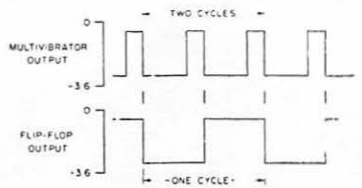


Fig. 15-22. Output of flip-flop.

Integrated Circuit Data

The integrated circuits used are manufactured by Fairchild Semiconductor, 313 Fairchild Drive, Mountain View, California 94040, and are available through their distributors.

These particular IC's are in a package with eight leads spaced on a 0.200" diameter circle protruding from the bottom of the package. There is a flat spot on the outer circumference of the unit. This flat spot is adjacent to pin 8. The remaining pins are numbered counterclockwise looking at the top of the package. The ICs are designed for a 3.6 volts DC plus or minus 10 percent supply (pin 8 positive, pin 4 negative). The manufacturer lists their operating temperature range as 15 to 55 degrees C. (Fig. 15-22)

Component Boards

Each of the three component boards is made from a 3" x 4" x 3-32" piece of micarta or phenolic. Brass eyelets 0.087" O.D. x 1/8" long are used for tie points.

The brass eyelets are inserted into all of the No. 43 holes in the component boards from the component side. Make certain the eyelets are pushed all the way into the board, so that the eyelet head is against the surface of the board. Turn the component board over, lay it on a piece of wood and funnel out each of the protruding eyelet "barrels" with a few gentle taps of a hammer on a 3/8" center punch. GC Electronics No. 7251 eyelets and a 3/8" punch identified as "PROTO 41" may be used.

All wiring is done on the back of the component boards in point-to-point fashion with No. 22 AWG tinned bus-bar wire. Insert the wires through the eyelets and bend the ends of each wire over on the component side of the board to hold the wire in place. Clip each wire next to the eyelets on the component

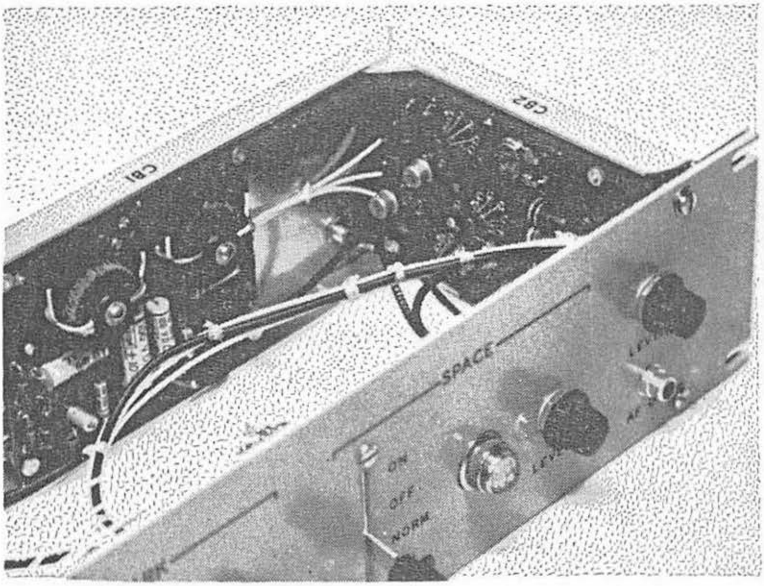


Fig. 15-23. Oscillator and amplifier lamp-driver boards shown installed. The power distribution board is mounted on the other end of the chassis.

side. Insert the components and solder each eyelet from the wiring side. Clipping off the excess component leads completes the component board wiring. The leads of all the semiconductor components should be heat sinked during the soldering operation.

Checkout and Adjustments

After the construction phase is complete, two pairs of electrical adjustments are required to place the unit in service;

1. R11 and R12 must be set so that the gates operate as amplifiers.
2. L1 and L2 must be adjusted (turns removed) to set the exact mark and space operating frequencies.

Connect a 600-ohm (nominal) load and oscilloscope to J3 and set R11 and R12 (on CB2) to the approximate center of their range. Apply power to the unit and check the minus 3.6 and minus 8.2 voltage levels. Set the mark and space amplitude controls (R9 and R10) to about mid-range and the AF output level control (R15) fully clockwise.

Place the mark keying selector switch (S4) to on and the space keying selector switch (S3) to off. DS1 should be illuminated and DS2 extinguished. Set the mark amplifier bias by adjusting R11 for maximum amplitude of the mark signal as displayed on the oscilloscope. Maximum amplitude and minimum distortion occur simultaneously.

Place the mark keying selector switch to off and the space keying selector switch to on. DS1 should be extinguished and DS2 illuminated. Adjust R12 for maximum space signal amplitude as observed on the oscilloscope.

Set the internal-external keying switch (S2) to external. Key the unit at J1 or J4 and check each of the possible keying combinations listed in Fig. 15-21. The responses of DS1 and DS2 should follow the signal output. Place S2 to internal and observe that the internal keying signal keys the unit at about a 22-Hz rate. The sync output signal at J2 should be a 22-Hz square wave at this time. This frequency has no particular significance other than being at about the same rate as the keying frequency of a 60 wpm speed machine. If it is too far off, bring it in by changing the value of R18. Bear in mind that each different set of components will have its own frequency vs R18 characteristics.

Because of the capacitor and inductor tolerances, it is extremely unlikely that the mark and space frequencies will be correct. The frequencies will probably be too low, but can be set to within a few cycles by removing turns from L1 and L2. Go easy here: it is a lot easier to keep on removing turns than it is to start adding them back. The frequencies may either be compared with an accurate audio (or AFSK) oscillator or measured with a frequency counter. The mark signal should be at 2125 Hz and the space signal at 2975 Hz. Soldering iron heat conducted to C1, C2, C4, or C5 will affect the oscillator frequencies. Frequency measurements should be made only after the capacitor temperatures have stabilized.

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